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Subject: Project Monitor at	25X1
d) indicated that he would soon submit a letter outlining:	25 <b>X</b> 1
1. The proposed new hardware delivery under Task 6 for the current funds in the original proposal.	
2. A cost estimate for carrying out several additional jobs under Task 6 (i.e. further work on the MC-20 and MC-30 and a complete final report of the work done under Task 3 and 6).	
e) RCA indicated that they had revised their thinking about the feasibility of cutting a message into an existing LP record. Pending the results of some tests to be done on their own they did not think that there was too much of a chance of accomplishing the desired objective.	
said that will soon submit:	25 <b>X</b> 1
a) A proposal for a research program on the solid state microphone.	
b) A proposal (about two man years of effort) to cover investigations of concealment techniques on microphone performance. This would also include a good deal of the binaural work outlined in a previous memorandum.	25X1
	25 <b>X</b> 1
TSS/APD	
Distribution:  P-189 - Orig.  P-119B - 1  P-185 - 1  Chrono - 1  TSS/APD/SPK: vs (2 May 1958)	





## CONFIDENTIAL

Final Report
Covering the Period
July 1, 1956 to Feb. 15, 1958

Report Date: March 15, 1958

#### AUDIO NOISE REDUCTION CIRCUIT

CONTRACT NO. RD-94 TASK NO. 2

25X1	Work Done and Report Prepared by:	
		25 <b>X</b> 1
	CONFIDENTIAL	

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#### **ABSTRACT**

A 110 channel audio noise reduction circuit has been built to cover the frequency range of 160 to 3260 cps. Each channel contains a tuned circuit and a non-linear device. The tuned circuit selects a narrow frequency range, and the non-linear device allows signals in this frequency range to pass whenever they exceed an arbitrary threshold level. The device performs the design functions satisfactorily, that is, it passes signals above a threshold level and rejects signals below this level. The pass to rejection ratio is about 40 db. When the system is used on test samples in which the speech is just intelligible in noise, the speech components which are greater than the noise can be passed virtually free from noise. However, these speech components appear as short tone bursts and do not provide intelligence. When the level of the composite signal is raised so that some noise is passed with the speech, the output signal is intelligible. The opinion of several listeners is that the noise reducer is not very effective in improving the intelligibility of speech in noise. There are times when the noise reducer appears to improve the intelligibility, but the results obtained so far are unsatisfactory and somewhat inconclusive. There are still many circuit variations that can be made before a final evaluation is possible.



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#### INTRODUCTION

The object of the work under this contract is to develop a noise reduction circuit which is capable of separating speech intelligence from a signal containing speech and noise in which the speech intelligence is masked by the noise. The method proposed utilizes a principle which has been used successfully to improve the signal-to-noise ratio in music reproducing and transmission systems. <sup>1,2</sup> In this method the signal is passed through one or more parallel channels formed by band pass filters which are less than one octave in width. These filters are used at the input and the output of a non-linear element. Although the output of the non-linear element contains the desired fundamental frequency and its undesired harmonics, the filters at the output of the non-linear elements remove the harmonics and pass only the desired fundamental frequencies.

The function of the non-linear elements is to reject all signals below a given amplitude or threshold level. The threshold levels of the non-linear devices in each channel can be adjusted so that, in the absence of the desired signal, all noise signals are rejected. When the desired signal is greater than the threshold level, the non-linear devices allow the composite signal to pass. Thus, for passages of recorded music when the music signal is below the noise level in a given channel, that channel is inoperative, and its output is eliminated from the total output. Since the contribution of this channel to the total output would have been only noise, the overall noise level is reduced. When the music signal in a given channel is greater than the noise, the channel conducts and allows the composite signal to pass. Thus, a channel conducts only when the desired signal is greater than the noise and rejects when noise alone is present.

In order to apply this method of noise reduction to speech signals in which the wide band signal-to-noise ratio is very low, it is necessary to find frequency regions in which, at times, the speech amplitude is greater than the noise. It was believed that this was possible because of the formant structure of speech. Although the long time average spectrum of speech is continuous and similar in shape to the spectrum of room noise, 3 the short time spectrum of various speech sounds contains regions of maximum energy called speech formants. Furthermore, these regions contain only the harmonics of the fundamental pitch of the speaker. For example, if the fundamental pitch of a speaker's voice for a given speech entity were 100 cycles per second, then the short time spectrum

<sup>1.</sup> Olson, H. F., "Audio Noise Reduction Circuits", Electronics, Dec., 1947.

<sup>2.</sup> Olson, H. F., Acoustical Engineering, D. Van Nostrand Co., Inc., Princeton, N. J. 1957, p. 420

Fletcher, H., Speech and Hearing in Communication, D. Van Nostrand Co., Inc. New York, 1953. (See Figures 61 and 70)

of this speech sound would contain harmonics that are multiples of 100 cps. The belief that a multi-channel threshold type noise reducer could be utilized for speech was based upon the assumption that it would be possible to locate frequency bands in which the amplitude of the speech formants was greater than the noise a substantial part of the time.

The problem involves the solution of the transient response of narrow band non-linear systems. It is complicated by the fact that the transients are not periodic but have the random distribution characterized by both speech and noise. It is necessary that the output of these systems be more than just detected. The output must be intelligible. Although some theories have been proposed to aid our understanding of the speech perception process, <sup>4</sup> the actual information bearing portions of speech are not clearly understood. Formal analysis of the problem appears impractical, so basicly, it has been studied on an experimental basis.

In attacking the problem the aim was: first, to determine the most desirable channel bandwidths to separate speech from noise; second, to obtain a satisfactory threshold device; and finally, to design the various circuit details. The basic design philosophy was to develop a simple, straightforward circuit which would prove whether or not this method of noise reduction could be applied to speech masked by continuous spectrum noise. Small physical size and compactness of design were not of primary importance. Therefore, no attempt was made to use sophisticated circuitry.

<sup>4.</sup> Licklider, J. C. R., "On the Process of Speech Perception," Jour. Acous. Soc. Amer., Vol. 24, No. 6, November 1952, p. 590.

#### DESIGN CONSIDERATIONS

In order to apply the threshold method of noise reduction to speech when the wide band speech signal-to-noise ratio is very low, it was first necessary to make a study to determine what bandwidths were necessary in order to have the amplitude of the speech components greater than the noise. It is known, that for noise with a continuous spectrum, it is the noise components in the immediate frequency region of the masked tone which contribute to the masking. When wide band continuous spectrum noise, which has been passed through a narrow band filter, is used to mask a pure tone centered in this filter, the masking increases as the bandwidth is increased until a certain bandwidth is reached After this, as the bandwidth of the masking noise is increased, the amount of masking remains constant. This bandwidth at which the masking reaches a fixed value is termed the critical bandwidth. The critical bandwidth is a function of frequency, and it is different when listening with one or two ears. The critical bandwidths for one and two ears as a function of frequency are shown by curves A and B in Figure 1. These bands are a subjective concept. Their actual shape is not known, but they are much narrower than the resonance curves of the cochlear partition as measured by Bekesy.

From the critical band concept, it is evident that the channel bandwidths of a noise reducer circuit must be narrower than the critical bands of the ear in order to effect an improvement in the signal-to-noise ratio as detected by the ear. The critical bands of the ear are quite narrow; therefore, a large number of bands will be needed in the noise reducer circuit.

The critical band concept indicates, that in order to detect a continuous single frequency signal in noise, the narrowest bands physically possible will give the greatest signal to-noise improvement. However, speech is a transient phenomena; and although the energy is concentrated in the vowels and formant structure, much of the intelligence is contained in the manner in which the vowels are started and stopped, i.e., in the consonants. If the channels of the device are too narrow, the consonant information will be deteriorated or eliminated

<sup>5.</sup> Fletcher, H., Speech and Hearing in Communication, D. Van Nostrand Co., Inc. New York, 1953, p. 171.

<sup>6.</sup> French and Steinberg, "Factor Governing the Intelligibility of Speech Sounds," Jour. Acous. Soc. Amer., Vol. 19, No. 1, Jan. 1947 (See Figure 7).

<sup>7.</sup> Bekesy, George V., "Resonance and Decay at Points on the Cochlear Partition," Jour. Acous. Soc. Amer., Vol. 21, No. 3, May, 1949, p. 250 (See Figure 7).

What minimum bands can be tolerated? From Fletcher, 8 the average duration of the various speech sounds are as follows:

Vowels 219 to 351 millisec.
Stop Consonants 20 to 190 millisec.
Fricative Consonants 70 to 300 millisec.

These values show that to pass the shortest stop consonants the signal in the channels must build up appreciably in 20 ms. The voltage amplitude in a resonant system builds up to 1-1/e or 63%, of its final value in  $Q/\pi$  cycles. The rise time t required for this build up is given by:

$$t = \text{No. of Cycles } \cdot \frac{1}{\text{freq.}}$$
 (1)

Q can be determined from the expression: 9

$$Q = \frac{f}{\Delta f} \tag{2}$$

where f = resonant frequency of system,

 $\Delta f = f_2 - f_1,$ 

12 = half power frequency above resonance,

 $f_1$  = half power frequency below resonance.

Substituting  $Q/\pi$  for the number of cycles in equation 1, and making use of relation 2, gives,

$$t = \frac{f}{\Delta f \cdot \pi} \cdot \frac{1}{f} = \frac{1}{\Delta f \pi} \tag{3}$$

$$\Delta f = \frac{1}{\pi t} \tag{4}$$

Using the value t = 20 ms. gives  $\Delta f = 16$  cycles as the minimum bandwidth that could be used for speech channels. It has been assumed here that a channel which permits a

<sup>8.</sup> Fletcher, H., Speech and Hearing in Communication, D. Van Nostrand Co., Inc. New York, 1953, pp. 62, 64 and 66.

<sup>9.</sup> Arguimbau, Vacuum Tube Circuits, John Wiley & Sons, Inc., N.Y.C., 1948, p. 182

signal to build up to 63% of its final value in 20 ms. would not seriously deteriorate speech intelligence. The straight line labeled (D) in Figure 1 shows the minimum bands allowed based upon this calculation

Having established the maximum and minimum boundary conditions for the channel bandwidths, it was necessary to determine the frequency range over which the circuit would operate. From the Articulation Index concept and the extensive work of Fletcher, French and Steinberg, it has been shown that extending the frequency range of a communication system below 200 cps or above 6000 cps adds very little to the intelligibility of speech. They have shown that in the absence of noise this frequency region may be broken into many narrow bands each one of which adds a contribution to the total articulation index. The total of all these bands gives 100% articulation. Frequently this frequency region is divided into 20 bands in such a manner that each band adds an equal amount to the total Articulation Index. Since there are 20 bands, each band adds 5%. The bandwidths and frequency location of these 5% bands are shown by the short horizontal lines in Figure 1

In the first design of a noise reduction circuit, the first four and the last four of these 5% bands were eliminated. This first circuit contained 80 channels and covered the frequency range from 700 to 3260 cps shown on Figure 1. This frequency region should yield an articulation index of 60% in the absence of noise. Later more channels were added to the system to cover the frequency range from 160 to 3260. This was done in order to increase the naturalness of the speech. In this final form the maximum possible articulation index is 80% in the absence of noise.

Utilizing the design considerations as outlined, the final noise reducer covers the frequency range from 160 to 3260 cps. The bandwidths of each channel were made 3 db narrower than the critical bands and spaced so that they crossed each other at the ½ power frequencies. These bandwidths were chosen because they should give detectable improvement if the system performed satisfactorily and should not seriously deteriorate the speech intelligibility. Referring to Figure 1, the bandwidths of the noise reducer were chosen to follow curve C. This curve is 3 db below the critical bandwidths for two ears, curve B; but it is above the minimum bandwidth for a 20 ms rise time, curve D. In this report bandwidth is frequently expressed in decibels above a one cycle band. This is defined as 10 log BW, where BW is expressed in cps.

French and Steinberg, "Factors Governing the Intelligibility of Speech Sounds," Jour. Acous. Soc. Ames., Vol. 19, No. 1, Jan. 1947, p. 97.

In order to determine the signal-to-noise ratio which could be expected if channel bandwidths approximating the critical bands were used, a series of experimental measurements were made using filters whose bandwidths were in the neighborhood of the critical bands. Studies were made of low signal-to-noise ratio samples of both pure tones and speech mixed with continuous spectrum type noises. The results of this study showed that, for the narrowest permissible bands which could be used to pass speech, the number of times the speech amplitudes, in a given band, exceeded the noise was small. Furthermore, in these bands, the speech amplitude was never considerably greater than the noise. Since only a small difference in amplitude exists between the desired signal and noise, it was concluded that the non-linear threshold device must have a sharp cut-off characteristic in order to reject the noise and pass the speech. This was in line with the original aim of the project; namely, to build a device which would produce speech free from noise. In retrospect it may be more desirable to have a non-linear characteristic similar to an expander in which the noise and speech are both passed, but the signal-to-noise ratio is improved.

The non-linear portion of the circuit was designed to have the sharpest rise or switching action that could be practically obtained. This was done so that speech could be separated from the noise when the speech amplitude did not greatly differ from the noise.

It would be difficult to elaborate all of the decisions which influenced the total design of the circuit. Each section was tailored to do the best possible job compatible with a practical system. The circuit was made as flexible as possible so that many different characteristics could be tried.

<sup>11.</sup> Davenport and Root, An Introduction to the Theory of Random Signals and Noise, McGraw Hill Book Co., Inc., New York, 1958, Chapters 12 and 13.

#### CIRCUIT DESIGN AND DESCRIPTION

In order to design a circuit which satisfactorily performs the functions outlined in the previous section, an answer must be obtained to the engineering question, "How close must the practical circuit appoximate the ideal?" In view of the large number of channels required, considerable weight must be given to circuit designs which minimize the number of tubes, components, and the power required. The design of the circuit can be divided into three main categories; channel filter design, non-linear circuit design, and the associated circuit design. These three categories will be discussed separately, and then the over-all circuit characteristics will be covered.

#### 1. CHANNEL INPUT FILTERS

Ideally each channel requires two narrow band pass filters, one at the input and one at the output of a non-linear device. The frequency response characteristic of an ideal filter is shown by the dashed curve in Figure 2. It has zero attenuation in the pass band and infinite attenuation at frequencies above and below the pass band. Multi-element filters can be designed to approximate this characteristic, but in this application over two hundred filters of very narrow bandwidth are required. If electrical band pass filters of the type shown in Figure 10 were designed, the physical size and cost would be impractical. For example, a 20 cps wide, band pass filter at a center frequency of 1000 cps and an impedance of 1000 ohms would require nine high Q inductors, four of which would have inductance values of 20 henries. It is possible that the physical size of the filters could be reduced by making use of a heterodyne system and incorporating electromechanical filters. However, there are no commercially available filters which would be satisfactory. 12 It does appear that the desired bandwidth could be obtained at a frequency of 100 kc by using specially built electromechanical filters. 13 These types of filters were considered too complex for the channel filters of the first noise reducer. If the threshold noise reducer principle proves to be valuable using simple filters, then more nearly ideal filters can be developed to improve its performance.

Lungren, D. L., "Electromechanical Filters for Single Sideband Application," Proc. IRE, December, 1956, Vol. 44, p. 1744.

<sup>13.</sup> George, R. W., "Electromechanical Filters for 100 kc Carrier and Sideband Selection," Proc. IRE, January, 1956, Vol. 44, p. 14.

The simplest type of selective device is a single resonant circuit. It has a minimum number of components and has the advantage that its bandwidth can be changed simply by varying the damping on it. Although the attenuation considerably off resonance increases only 6 db per octave, it will be shown later that the characteristic of the non-linear device has a large effect upon the effective channel bandwidth. When the response of the tuned circuit is added to the characteristic of the non-linear element, the ideal filter response curve is approximated at levels close to the threshold.

Before the actual components required for these channel input filters can be considered, some knowledge of the bandwidth and band spacing will be required. From Figure 1 it is apparent that filters 3 db narrower than the critical bands of the ear should provide a satisfactory transient response for speech, and yet give better noise rejection than the ear. Therefore, the bandwidths of the channels were initially chosen to fit the -3 db curve To facilitate band spacing and bandwidth adjustment, the bandwidths were quantized to the discrete values shown by the short solid lines on Figure 3. The frequency range of the 110 channel circuit was chosen from 160 to 3260 cps. The design values of band center and bandwidth are given in Table I.

In order to approximate the bandwidth of the channels with simple resonant circuits, the effective bandwidth of the response curve must be ascertained. The universal resonance curve for a parallel tuned circuit is shown by the solid curve in Figure 2.  $^{14}$  The effective or integrated bandwidth is shown by the dashed curve. These two curves pass equal amounts of noise power. The effective bandwidth of the universal resonance curve is 1.44 times the bandwidth of this curve measured at the half power frequencies. Stated inversely, the bandwidth at the half power points must be 0.7 of the desired bandwidth. For example, at 800 cps an effective bandwidth of 20 cps is required in order that the channel be 3 db below the critical band for two ears. To obtain this effective channel width, the bandwidth of the resonant circuit at the half power frequencies must be 0.7 x 20 or 14 cps. The Q of the resonant circuit from expression 2 is  $800 \div 14 = 57$ . A curve of the Q's required for the tuned circuits as a function of frequency is given in Figure 4.

The maximum Q of the tuned circuits will, in general, be determined by the Q of the inductors. The Q of the inductors should be substantially above the maximum circuit Q required. The maximum Q of the various inductors are shown in Figure 4

In addition to high Q the value of the inductance should be stable with respect to a wide range of applied voltage, temperature, frequency and time. Rather than design and

<sup>14.</sup> Terman, F. E., Radio Engineers' Hand Book, McGraw Hill Book Company, Inc., N.Y.C., 1943, p. 137.

wind components to meet the requirements it was more convenient to purchase stock items UTC type M toroids were found to be satisfactory and have been used in the 110 channels. The design value of the reactance of the inductors was 1000 ohms. The actual values of the reactances are given in Figure 5. The impedances of the tuned circuits at resonance are a function of the damping on the circuit; i. e., Q times the reactance.

The capacitors used to resonate the inductors must be as stable as the inductors. Aerovox type L84 polystyrene capacitors fulfilled this requirement. The choice of compoents has proved to be satisfactory. The center values were adjusted to within  $\pm$  1 cps using an electronic counter. After three weeks the frequencies of the channels were found to be within  $\pm$  3 cps of the design values. The design values of the inductors and capacitors are shown in Table I (Columns L and C).

The shape and overlapping of the response frequency characteristics of a small group of these channels when the tuned circuit had an equivalent bandwidth 3 db below critical are shown in the lower view of Figure 34. This figure covers channels 111 to 120 which are physically located on chassis \*2.

#### 2. OUTPUT FILTERS

The output filters that remove the distortions from the channel outputs caused by the non-linear devices are ideally the same as the input filters. However, it was believed that a single tuned circuit would not provide sufficient harmonic rejection. Two steps were taken to provide a sharp filter characteristic without resorting to a large number of high quality components. First, since these output filters are used to remove harmonic distortion, they have been designed to cover a group of channels over a frequency range of less than one octave. It has been found convenient from several points of view to arrange the channels in groups of ten on a single chassis. Second, since most of the harmonic distortion introduced by the non-linear elements is made up of the higher harmonics, it was possible to use low pass filters rather than band pass filters. The configuration used for the output filters is shown in Figure 6. The measured response frequency characteristic of each of the output filters is shown in Figures 7, 8, 9.

#### 3. CHASSIS BAND PASS FILTERS

In the original thinking on the design it was assumed that very little speech signal above threshold would be required in any particular channel. This was based upon the assumption that the threshold levels could be programmed for any given passage of speech to pick out only the speech information. If this were the case then the input filters described would be sufficient. However, it has been found that the noise reducer requires a signal whose peaks are from 15 to 20 db above threshold. When a signal of this amplitude is used with the single tuned circuit as a channel filter, 10 to 15 adjacent channels may pass the signal. In order to prevent signals from exciting adjacent chassis, eleven band pass filters have been built to be used before the channel filters. One filter was placed at the input of each chassis so that only 10 channels received a common signal. The band pass filter configuration of these filters is shown in Figure 10. The measured response frequency characteristics are shown in Figures 11, 12, 13 and 14. The effectiveness of these filters in isolating adjacent chassis is demonstrated by the two small vertical lines on each side of the response curve. The dashed line indicates the channels which are to be rejected. The solid line shows the channels which must be passed.

#### 4. NON LINEAR ELEMENTS

Equally important to the successful operation of the threshold noise reducer is a satisfactory non-linear element or threshold device. As a first step toward designing a threshold device, the physical and electrical characteristics of several diode elements were studied. 15, 16 High vacuum diodes, gas diodes, 17 and crystal diodes were considered. It appears that, from the standpoint of circuit simplicity, the germanium diodes are the most desirable; however, because of their relatively low reverse resistance, the electrical characteristic of the germanium diodes are not as satisfactory as the high vacuum diodes. A comparison of the input-output characteristic of a 6AL5 high vacuum diode and a 1N54A crystal diode is shown in Figure 15A and 15B. Figure 15A is a linear plot expressed in relative volts. Figure 15B is a logarithmic plot of the same data expressed in decibels. These are typical of the difference between high vacuum diodes and crystal diodes.

<sup>15.</sup> Chance, B. et. al., Waveforms, McGraw Hill, New York, 1949, Chapter 3.

<sup>16.</sup> Crystal Diodes, Sylvania Bulletin Ec. 36, Rev. 1-110-555.

<sup>17.</sup> Townsend, M. A. and Depp, W. A., "Cold Cathode Tubes for Transmission of Audio Signals," Bell Systems Tech. Jour., Vol. 32, November, 1953, pp. 1371-1391.

Originally, it was thought that the sharpest characteristic possible, i e, a perfect switch, would be ideal for the separation of speech from noise when there is little difference between the speech signal and the noise. This ideal characteristic for the non-linear device is shown in Figure 15C and 15D. The output changes abruptly when the input signal rises above the threshold, but then it continues linearly above that level. This type of characteristic can be approximated by a pair of biased diodes, as shown in Figure 16. In the final design a high vacuum diode was used for diode \*1 to give a sharp switching characteristic, but the other non-linear needs were filled with crystal diodes. The final design of the threshold device is shown in the single channel schematic of Figure 17. The input output characteristic of this circuit is shown in Figures 15E and 15F. This is not the characteristic for the total noise reducer, but rather only for the non-linear portion

#### 5. EFFECTIVE CHANNEL BANDWIDTH

The response frequency characteristic of the single channel circuit of Figure 17 is a function of the amplitude level of the signal above threshold as well as the Q of the tuned circuit. Figure 18 is an attempt to portray the combined effect of the tuned circuit and the non-linear device at various levels above threshold. Figure 18A shows the response characteristic of the tuned circuit at the input of one channel. The output of this tuned circuit is the input to the non-linear element. Figure 18B is the input output characteristic of an ideal non-linear element. Figure 18C shows the response frequency characteristic of combination of 18A and 18B. Three different level are shown. Curve 1 shows a tuned circuit output which is 5 db above the threshold level of the non-linear element. It can be seen that the combined response frequency characteristic for this level Figure 18C is quite close to the ideal band pass filter characteristic. Curve 2 shows the effect for the signal about 12 db above threshold and Curve 3 shows the effect about 20 db above threshold. It can be seen that the bandwidth of the curves in Figure 18C increases in width as the level increases; and furthermore, even thought the shapes of the three curves in Figure 18A are identical, their combined shape (Figure 18C) changes from an ideal one at very low levels to the universal resonance shape at high levels. This type of combined effect will always exist as long as the channel filter characteristic has a finite slope between the pass and rejection regions This variable bandwidth effect is believed to be a desirable characteristic because it provides a very narrow band and good noise rejection for very low signals, but passes a wider range of frequencies when the desired signal is well above the noise

In all of the bandwidth adjustments and measurements made, the signal was adjusted to be 3 db above threshold at the resonant frequency of the tuned circuit Response frequency measurements of the various channels and their effective bandwidths were determined for this level. The lower part of Figure 32 shows the effective bandwidths of channels 111 to 120 measured with a signal 3 db above threshold. The channel bandwidths in this figure are 3 db narrower than the critical band. The response frequency characteristics of the tuned circuits alone are shown in the upper part of Figure 32.

In addition to the filters and the non-linear elements, several other circuit designs were developed to build the complete noise reducer. These will be described separately, and then the complete noise reducer circuit will be described.

#### 6 SINGLE CHANNEL CIRCUIT

In view of the large number of channels required, considerable thought was given to a single channel circuit which would perform satisfactorily and yet require a minimum of tubes, components, and power. The single channel circuit which was shown in Figure 17 seemed to fit these requirements. It utilizes one glass envelope per channel with a filament current of 0.3 amperes and a plate current of 1.0 milliamperes.

#### 7. VOLUME COMPRESSION CIRCUIT

A fast acting volume compression circuit has been used on each chassis to reduce the amplitude range of the signal. The compression circuit was placed immediately following the input band pass filter. The input-output characteristic of a typical volume compressor is shown in Figure 19. A circuit schematic is shown in Figure 20.

#### 8. INPUT-OUTPUT AND BY-PASS CIRCUIT

In order to test the effectiveness of the threshold noise reducer, a circuit was built which has the same total bandwidth as the noise reducer. By switching from the noise reducer to this circuit, a direct comparison can be made of the signal through the noise reducer and then through the by-pass. This type of A-B test has been used for evaluation throughout the project. The switch, an input amplifier, an output amplifier and the by-pass circuit of the noise reducer are all placed on the input-output chassis. A circuit schematic of the input-output chassis is shown in Figure 21. The response frequency characteristic of the by-pass circuit is shown in Figure 22.

#### 9. GENERAL CIRCUIT DESCRIPTION

The total noise reducer circuit is made up of the various sub-circuits that have been previously described. For several reasons it has been found convenient to arrange the channels in groups of ten and place them on one chassis. It facilitates assembly, simplifies the output filtering, makes possible volume compression for the groups, and reduces the number of threshold adjustments. A block diagram of one chassis is shown in Figure 23. The circuit is an assembly of the sub-circuits. At the left is a band pass filter, details of which are given in Figure 10. Next, moving from the left, is the volume compression circuit; it is detailed in Figure 20. Following a driving amplifier are ten single-channel circuits, each containing a tuned circuit, amplifier and a non-linear section. The outputs of the ten channels are mixed together and then fed to a second amplifier and non-linear element which operates upon the combined signals from the ten channels. Therefore, the signal from any one channel passes through two non-linear elements in series. This was needed to reduce the leakage noise through each channel. The second non-linear element greatly reduces the output noise in the absence of signal. Following the second non-linear element is an amplifier, the low pass filter and finally an output amplifier. The low pass filter configuration is shown in Figure 6.

Three overall input-output characteristics for some representative channels are shown in Figure 24. The compression action was set to act at the threshold of the non-linear elements. Figure 25 shows the effect of various volume compression thresholds. The dynamic range of speech covers about 30 db. <sup>18</sup> It can be seen that the circuit begins to clip after about 10 db above threshold, but it is believed that this does not seriously affect the intelligibility of the speech. <sup>19</sup>

A schematic diagram of one chassis is shown in Figure 26. Photographs of a typical chassis are shown in Figure 27.

Eleven of these chassis are used to form the complete 110 channel noise reducer. A block diagram of the complete system and the test conditions are shown in Figure 28. The input-output and by-pass circuits have been described in Figure 21. Photographs of the complete system are shown in Figures 29 and 30. No attempt has been made to show a

<sup>18.</sup> Dunn and White, "Statistical Measurement of Conversational Speech," Jour. Acous. Soc. Amer., Vol. 11, 1940, p. 278

<sup>19.</sup> Licklider and Pollack, "Effects of Differentiation, Integration, and Infinite Peak Clipping Upon the Intelligibility of Speech," Jour. Acous. Soc. Amer., Vol. 20, No. 1, Jan. 1948, p. 42.

complete schematic. Table I gives the values of various components on each chassis. The chassis are numbered from 1A to 8. The channels have been designated from 71 to 180. Response frequency characteristics of the complete noise reducer are shown in Figure 31. The upper two characteristics are for bandwidths of -6db; the lower two characteristics are for bandwidths of -3 db. The characteristics are shown for signals 10 db above threshold and 20 db above threshold.

#### CIRCUIT OPERATION AND TESTS

The threshold noise reduction circuit is a very flexible device which has a great many possible adjustments. For example, the following adjustments can be made using external controls

- 1. Channel Bandwidths
- 2 Non-linear Element Slopes
- 3. Relative Channel Thresholds
- 4 Volume Compression Characteristics

This means that 24 combinations can be used and each of these combinations has a continuously variable adjustment. In addition, various types of test samples can be used for any given noise reducer condition. At present only a relatively few variations have been tried. These are the channel bandwidths, the number of channels, and relative channel threshold. The number of channels was either 80 or 110. The first noise reducer tested had 80 channels; later, after some tests, the number of channels was increased to 110.

#### 1 CHANNEL BANDWIDTHS

Two channel bandwidths have been used on the noise reducer: one 3 db narrower than the critical bandwidth, and one 6 db narrower than the critical bandwidth. The design values of these bandwidths are shown by the horizontal straight lines in Figure 3 and are tabulated in Table I. The deviation from the design values of the actual bandwidths of the individual channels are shown by the points in Figure 3. These were measured after the circuit had been used for a few weeks. Frequency was measured using an electronic counter. The response frequency characteristics of ten adjacent channels are shown in Figures 32, 33 and 34. Figure 32 is for bands 3 db narrower than critical. Figure 33 is for bands 6 db below critical. The upper set in each figure shows the response of the tuned circuits, the lower set shows the effective bandwidth 3 db above threshold. This effective bandwidth has been described. See Figure 18. Figure 34 shows the tuned circuit response which gives 3 db and 6 db bandwidths neglecting the effect of the non-linear elements.

#### 2. CHANNEL THRESHOLD SETTINGS

Two types of channel threshold setting have been used, one in which all channel thresholds were adjusted equal and one in which groups of ten channels were adjusted to follow the shape of the spectrum of speech. These two characteristics are shown in Figure 35

#### 3. VOLUME COMPRESSION CHARACTERISTIC

Only one volume compression characteristic has been used. However, the compression point was set at either 0 or 5 db above threshold; i. e., the compressor did not act until the signal was either at threshold or 5 db above threshold. The input-output characteristic of a typical volume compressor is shown in Figure 19.

#### 4. NON-LINEAR CHARACTERISTIC

Only one non-linear slope characteristic has been used. It was the maximum slope which could be obtained and maintain a 30 db dynamic range for the driving amplifier in each channel. The bias for this slope is 5 volts. This slope is shown in Figure 15F.

#### 5. TEST SAMPLES

Three test samples were used to measure the effectiveness of the noise reducer. These samples were: speech without noise, speech mixed with various level of flat noise, and speech mixed with various levels of simulated room noise. The spectrum of these two types of noise are shown in Figure 36.

A summary of the various circuit conditions used up to the present is given in Table II.

#### 6. EVALUATION METHOD

The test procedure used for evaluating the noise reducer under the various conditions is as follows: A few paragraphs of speech of nearly constant intensity were recorded on tape. These recordings were then mixed with either the flat or the simulated room noise. Several different signal to noise ratios were used, but usually the noise level was raised until the threshold of intelligibility was reached <sup>20</sup> The adjustment was made subjectively by

<sup>20.</sup> Hawkins and Stevens, "The Making of Pure Tones and of Speech by White Noise," Jour. Acous. Soc. Amer., Vol. 22, No. 1, Jan. 1950, p. 11.

listening and raising the noise level until the listener could just understand the context of the sentences. It was found that three independent observers set the signal to noise ratio within ± 2 db. This adjustment was made with the composite signal passed through the by-pass circuit. After the signal-to-noise ratio was adjusted through the by-pass, the composite signal was switched through the noise reducer circuit at various levels above threshold, and the observer was asked to indicate whether or not the noise reducer improved the intelligibility of the speech. The results of the various tests will be discussed in the next section. The test used here gives merely a qualitative result of the value of the noise reducer.

There are many types of speech samples and test procedures that have been used by other experimenters to establish quantitative values of the articulation index. <sup>21</sup> For example, experimenters have used nonsense syllables, monosyllabic words, phonetically balanced or P. B. lists, dissyllabic words, spondee lists, sentences, and continuous discourse. Although intelligibility could be measured, quantitatively, by using one of these standard articulation tests, these tests require trained talker-listener groups and are very time consuming. It is felt that satisfactory qualitative evaluation of the various circuit conditions can be obtained from the simple comparison tests. If a promising circuit condition is obtained, then one of the standard articulation tests can be used to obtain a quantitative value.

#### VOLUME COMPRESSOR EVALUATION

A special test was performed to determine the effect of the eleven volume compressors and the band pass filters. In this test,  $V_{12}$  was removed from ech chassis and a jumper was connected from  $J_{1F}$  to  $J_{11F}$  on each chassis, (See Figure 26). This passes the speech through the band pass filters, the volume compressors and the output low pass filters, but it eliminates all of the narrow channel filters and the non-linear elements. Speech passed through this portion of the circuit did not lose any appreciable amount of intelligibility; therefore, it was assumed that the overlapping of the band pass filters, the phase shift variations on different chassis, and the time constants of the volume compressors were satisfactory.

<sup>21.</sup> Hirsh, I. J., The Measurement of Hearing, McGraw Hill Book Company, Inc., New York, 1952. Chapter 5

#### RESULTS AND CONCLUSIONS

A multi-channel threshold noise reduction circuit has been built which, in its final form, covers the frequency range from 160 to 3260 cps. Several different circuit conditions and test samples have been used to evaluate the effectiveness of the device. The results of these tests can be summarized in two classes: first, the noise reducer with channel bandwidths 3 db narrower than the critical bandwidths of the ears; and second, the noise reducer with channel bandwidths 6 db narrower than the critical bands.

Consider first the results of tests on the noise reducer with -3 db bandwidths Although the final proof of the effectiveness of the device requires an auditory evaluation, visual observations help to give some insight into the performance of the system. Oscillograms have been made for several different test conditions. These oscillograms are shown in Figures 37 to 42.

In all of these figures the top view, A, shows a given speech sample without noise. View B is this same speech sample mixed with an amount of noise that makes the composite signal just intelligible. In both of these views the signal had been passed through the bypass circuit which has the same bandwidths as the noise reducer. (See Figure 22 for the response-frequency characteristic of by-pass.) Views C. D and E were made with the composite signal passed through the noise reducer at different levels above threshold. (See Figure 31 for the response-frequency characteristic of the noise reducer.) View C is taken with the signal set at the maximum level which just excludes noise. View C visually resembles the speech in A closer than it does the composite signal of B. Views D and E show the effect of greater amounts of signal above the noise reducer threshold. It can be seen that E approaches B.

There are four figures shown for the -3 db bandwidths. The first two, Figures 37 and 38, were made with the noise reducer thresholds set flat. (Refer to Figure 35.) In Figure 37 the speech signal was mixed with flat noise. In Figure 38 the speech was mixed with room noise. In the next two figures the noise reducer thresholds are adjusted to follow the shape of room noise. (Refer to Figure 35 for this characteristic.) Figure 39 is for flat noise; Figure 40 is for room noise. All of these oscillograms show that there are some speech signals greater than the noise, and that the noise reducer can separate these signals from the noise

When these signals are judged aurally, the signal displayed by view C is unintelligible. This is true even though the signals of view A and B are intelligible A signal as shown in view D gives some intelligence and view E is judged most intelligible In general, a signal as shown in view E is not judged to be more intelligible than a signal as depicted by view B

There are two conditions shown for the noise reducer with 6 db bandwidths. These were made with the noise reducer thresholds set flat Figure 41 is for flat noise; Figure 42 is for room noise. When the noise reducer with 6 db bandwidths was evaluated aurally, it was less intelligible than the circuit with 3 db bandwidths. Actually, it was found that the intelligibility of a pure speech signal was seriously impaired when it was passed through the 6 db bandwidth noise reducer. The cause of this deterioration has not been completely determined. Using these very narrow bands with the band spacing of Table I has two effects First because the filters are very narrow, the transient response is not satisfactory for speech. Second, with the band spacing used, the 6 db bandwidths reject 50% of the speech information. Either of these could seriously impair the speech intelligibility, no attempt has been made to separate the two effects

The conclusion reached from all of the tests made is that the threshold noise reducer is capable of separating some speech components from a composite signal in which speech is just intelligible in noise. These speech components are passed relatively free from noise. The intelligibility of the speech formed by these components is very small, if not zero When the signal level is increased so that a large amount of noise is passed with the speech, the noise reducer provides very little improvement. What improvement there is seems to be partially masked by the switching action or "birdies" caused by the non-linear elements

The most effective results were obtained with the noise reducer bandwidths 3 db narrower than the critical bands of the ears, the noise reducer thresholds set flat, and the speech masked with flat noise

#### **FUTURE WORK**

Although the effectiveness of the noise reducer in improving speech intelligibility has not been completely determined, it has shown enough promise to warrant further evaluation and modification. There are many variations that are yet to be tried

One of the most important of these is the characteristic of the non-linear element. The basic philosophy used throughout this project was to clip off the speech and reject the noise. It has been found that the speech information obtained in this manner does not provide intelligibility. It has been necessary to pass both speech and noise through the noise reducer in order to provide intelligibility. In view of this it is conceivable that a sharp non-linear characteristic is not the most desirable. Perhaps some other characteristic, such as a square law detector might prove more useful. This, of course, would not separate the speech from the noise but would improve the signal-to-noise ratio.

Several other variations can be tried. For example, further bandwidth variation, different channel threshold characteristics, and various amounts of volume compression are all possible useful variations.

	껊	kilo ohms							12 k				78 K	27 k	30 k			18 k		22 k	22 k					24 k		51 k		100 k	100 k	100 k	
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	ы	립		1000	1000	1000	1000	1000	1000	1000	1000	90	1000	1000	1000	1000	200	200	200	200	500	200	200	200	200	200	200	500	200	200	200	200	200
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	nel idth	edb	ф	9.55	9.55	9.55	9.55	9.30	9.30	9.30	9.30	9.30	9.30	9.30	9,30	9.30	9.30	9,30	9.55	9.55	9.55	9.55	9.55	5.5	9.55	15. Q	9.55	9.55	9.55	9.55	9.55	9.55	9.55
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	ч	녵		500	200	200	500	200	500	200	200	200	200	200	909	200	200	500	200	500	200	200	200	120	120	120	120	120	120	120	120	120	120
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T'D	Channel Bandwidth	edb	ą	10	10	10	10	10	01	10	10	10	10	10.42	10.42	10.42	10.42	10.42	10.42	10.42	10.42	10.42	10.42	10.8	10.8	10.8	10.8	10.8	10.8	10.8	11.15	11.15	11.15
I - CONT'D	Cha) Bandi	ı	<b>8</b> 000	10	10	20	10	10	10	10	2	, 10,	10	11	11	Ħ	11	Ħ	#	==	11	11	11	12	12	12	12	75	12	12	13	13	13
TABLE I	Channel Bandwidth	3db	qр	13	13	13	13	13	13	13	13	13	13	13.42	13.42	13.42	13,42	13.42	13.42	13.42	13.42	13.42	13.42	13.8	13.8	13.8	13.8	13.8	13.8	13.8	14.15	14.15	14.15
	Channel Bandwidt	1	<b>s</b> d၁	50	20	20	20	20	20	20	20	20	20	22	22	22	22	22	22	22	55	22	22	24	24	24	24	24	24	24	56	56	<b>56</b>
	Channel	Center	အ ဝ	700	720	740	760	780	800	820	840	998	880	006	922	944	996	988	1010	1032	1054	1076	1098	1120	1144	1168	1192	1216	1240	1264	1290	1316	1342
	Channel	No.		101	102	103	18	105	106	107	108	109	110	111	112	113	114	115	116	117	118	119	120	121	122	123	124	125	126	127	128	62 T	130
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	ا ئ ت		120	120	120	120	120	120	120	120	120	120	70	20	20	70	20	70	20	20	2	70	92	2	2	70	2	2	2	2	8	70
	Schematic Channel	•	ч	23	છ	4	വ	9	7	ထ	ത	10	ч	∾	8	4	വ	9	7	80	ക	10	н	∾	છ	4	S	9	7	œ	6	10
	Low Pass Filter	sdo chs					1700										2000										2400					
Γ¹D	nel idth	6ab ab	11.15	11.15	11.15	11.47	11.47	11.47	11.47	11.47	11.77	11.77	11.77	11.77	11.77	12.05	12.05	12.05	12.05	12.05	12.31	12.31	12.31	12.31	12.56	12.56	12.56	12,56	13.02	13.02	13.02	13.02
I - CONT'D		sďo 9 -	13	13	13	14	14	14	14	14	15	15	15	15	12	16	16	16	16	16	17	17	17	17	18	18	18	18	20	ଷ୍ଟ	20	20
TA BLE	Channel Bandwidth	3db db	14,15	14.15	14.15	14.47	14.47	14.47	14.47	14.47	14.77	14.77	14.77		14.77	15.05	15.05	15.05	15.05	15.05	15.31	15.31	15.31	15.31	15,56	15.56	15.56	15.56	16.02	16.02	16.02	16.02
	Cha: Band	São	26	26	58	28	28	82	28	82	30	30	30	30	30	3 6	325	32	32	32	34	34	34	34	36	36	36	36	40	40	40	40
	Channe 1	Center Cos	388	1394	1420	1448	1476	1504	1532	1560	1590	1620	1650	1680	2001	1742	1774	1806	1838	1870	1904	1938	1972	2008	2042	2078	2114	2150	2190	2230	2270	2310
	Channel	No.	131	130	1 to 20 C	134	135	136	137	138	139	140	141	142	143 143	144	145	146	147	148	149	150	151	152	153	154	155	156	157	158	159	160
	Chassis	• 0 12					4	•						23			LC.	ò									ç	)				

				TAI	TABLE I - CONT'D	ON I'D					
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	161	2350	40	16.02	20	13.02		1	50	.091	51 k
	162	2390	40	16.02	50	13.02		~2	20	.088	
	163	2430	40	16.02	20	13.02		ы	50	.085	51 k
	164	2470	40	16.02	20	13.02		4	20	.0825	
7	165	2514	44	16.43	22	13.43	2800	ഹ	20	.080	
•	166	2558	4	16.43	22	13.43		9	90	.077	
	167	2602	44	16,43	22	13,43		7	20	•074	
	168	2646	\$	16.43	22	13,43		80	20	.072	
	169	2690	4	16.43	22	13,43		6	20	0.00	
	170	2734	44	16.43	23	13,43		10	20	.067	
;	171	2782	48	16.81	24	13,31		Н	20	.065	75 k
24	172	2830	48	16,81	24	13,81		co.	20	.063	75 k
	173	2878	48	16.81	24	13,31		જ	50	.061	
	174	2926	48	16,31	24	13.31		4	20	•059	
00	175	2974	48	16,81	24	13.31	3600	ည	20	.057	
•	176	3026	52	17,16	56	14.16		ယ	20	.055	
	177	3078	52	17,16	26	14.16		7	20	<b>.</b> 053	
	178	3130	52	17.16	56	14.16		89	20	81	
	179	3182	52	17.16	56	14.16		G.	20	.049	
	180	3234	25	17.16	56	14.16		10	90	.048	100 k

TABLE II

SUMMARY OF TEST CONDITIONS

No. of Channels 80 110 110 110 110 80 80 80 Volume Compression Point -5db -5db -5db -5db Odb gg Odb Odbo Sloping Channel Threshold Sloping Sloping Sloping Flat Flat Flat Flat Non-linear Element Characteristic 5 volts Bias Max. Slope iiax. ax. wax. iax. ax. Max. Max. Effective Bandwidth -3db -3db qpgedb--3db -3db -6db -ನಿಡಿಶಿ

#### APPENDIX 1

#### **SPECIFICATIONS**

#### INPUT-OUTPUT CHASSIS

INPUT

Impedance = 600 ohms to transformer

Maximum AC = 0.15 volts
Maximum DC = 0 volts

OUTPUT TO NOISE REDUCER

Impedance = 600 ohms

Maximum AC = 17 volts to 1000 ohms

INPUT FROM NOISE REDUCER

Impedance = 2000 ohms

Maximum AC = 2.2 volts

Maximum DC = 0 volts

OUTPUT

Impedance = 600 ohms

Maximum AC = 20 volts (open circuit).

CONTROLS

Noise Reducer Gain

By-Pass Gain Output Level

#### TEN CHANNEL CHASSIS

INPUT

OUTPUT

 $\begin{array}{lll} Impedance & = & 5000 \ ohms \\ Maximum \ AC & = & 1 \ volt \end{array}$ 

CONTROLS

Input to Volume Compressor = P5FOutput from Volume Compressor = P1F

Channel Band Width = P71 to P180

Channel Gain = P71A to P99A, P201 to P280 Chassis Threshold Device Gain = P1A, P1B, P1C, P1 to P8

 $\begin{array}{lll} \text{Chassis Output Level} & = & P3F \\ \text{Channel Bias} & = & P2F \\ \text{Chassis Bias} & = & P4F \end{array}$ 

#### APPENDIX 2

#### ALIGNMENT PROCEDURE

- 1. Check all D. C voltages on chassis (See Appendix 3).
- 2. Tune channels to proper frequencies.

This requires a stable oscillator or signal generator, a frequency measuring device capable of reading frequency to ± 1 cps and an oscilloscope or a meter. Connect the oscilloscope or meter to the appropriate channel through the output jack on the rear of the chassis. These are numbered from J 71 to J 180. The design frequency value for any specific channel is given in Table I.

3. Adjust bandwidth of each channel.

The same equipment used to tune the channels can be used to adjust the bandwidths. The bandwidths of each channel can be controlled over a limited range by adjusting the variable resistors numbered from P 71 to P 180 located on the rear of the chassis. The procedure is as follows:

- a. Introduce a signal at the frequency of the channel center 3 db greater than the threshold.
- b. Adjust the variable resistor until the proper bandwidth is reached. This can be determined from the frequency of the half power points by using equation 2. If the value of bandwidth desired is outside of the range of the variable resistor, then the value of R (See Figure 26) must be changed
- 4 Adjust gain of each channel

Use the same equipment as outlined above.

- a Set signal on channel center frequency
- b Adjust the input level to the chassis so that approximately 1.0 volt appears on J1F (See Figure 26). This level should be below the value at which the volume compresor begins to act.
- c. Put scope on J11F
- d. Raise bias level, P2F, until the signal is just visible on the scope. This is the threshold level
- e. Adjust all other channel gains on this chassis for the same threshold, or as is desired.

#### 5. Adjust bias values.

In general, any value of bias voltage may be used; however, the shape of the input-output characteristic is a function of the bias level — the greater the bias voltage, the steeper the input-output characteristic. There are two bias adjustments on each chassis

- a. Channel bias adjustment
  - 1. Measure the D. C. bias voltage on J2F.
  - 2. Adjust P2F to the desired bias value. (Approximately 5 volts here gives a good cut-off characteristic and still provides 30 db dynamic range.)
- b. Chassis bias adjustment.
  - 1. Measure the D. C. bias voltage on J4F.
  - 2 Adjust P4F to the desired value. (It has been found convenient to set this bias voltage between 5 and 10 volts.)
  - 3. Adjust pots labeled (1A, 1B, 1C, 1 to 8, depending upon chassis) so that the channel signal must be 0.3 db above channel threshold before it exceeds the chassis threshold. (There are many variations possible here; a complete investigation of combinations has not been made.)

#### 6. Volume compression level.

The characteristic of the volume compressor is fixed for each chassis; however, the level at which it begins to operate can be adjusted relative to the threshold level.

To adjust the volume compressor to act 5 db above the threshold proceed as follows:

- a Set the chassis gain P1F to maximum.
- b. Place scope and Ballantine meter or equivalent on J1F.
- c. Increase the input level to the chassis until the volume compression point is reached. (See Figure 19 for volume compression characteristics.)
- d Place scope on J11 or J11F and adjust chassis gain P1F until signal just exceeds the threshold.

### APPENDIX 3

### DC VOLTAGES

(These are representative values)

Tube No.	Pin No.	Voltage Range Volts				
Tube No.	<u> </u>	¥0165				
1, 3, 5	1, 6	200 - 220				
7, 9, 12	2, 7	0				
, ,	3, 8	1.9 - 2.1				
	,					
11	1	0				
	2, 7	8 - 10				
	5	140 - 170				
	6	180 - 190				
14	1	180 - 200				
	2, 7	0				
	3	1.7 - 1.9				
	6	<b>16</b> 0				
	8	1.7				
4 P		110				
15	1	110				
	2, 7	0				
	3, 8	1.4				
	6	150				
16	1	0				
10	$\frac{1}{2}$	5 - 6				
	5	60				
	6	90				
	7	0				
	•	U				
17	1	.45				

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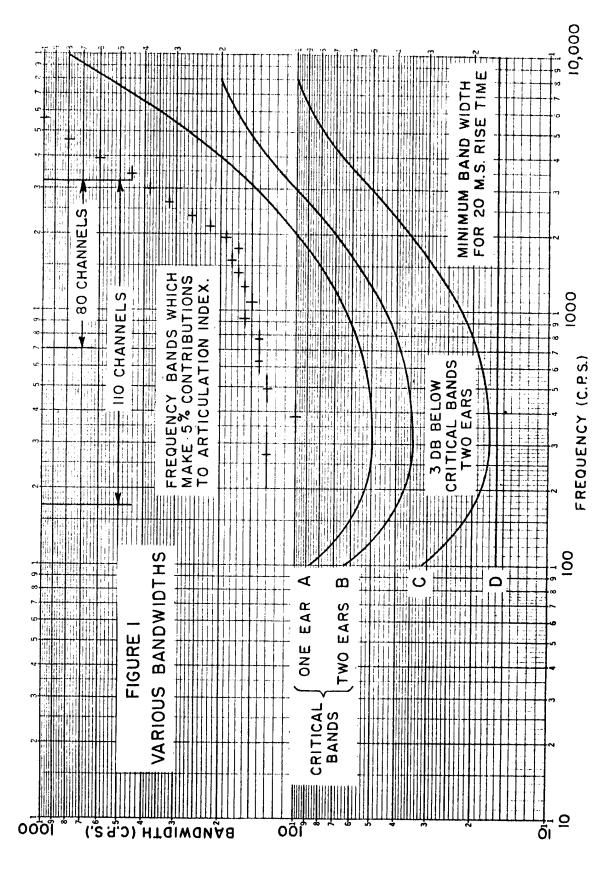
### **BOOKS AND MANUALS**

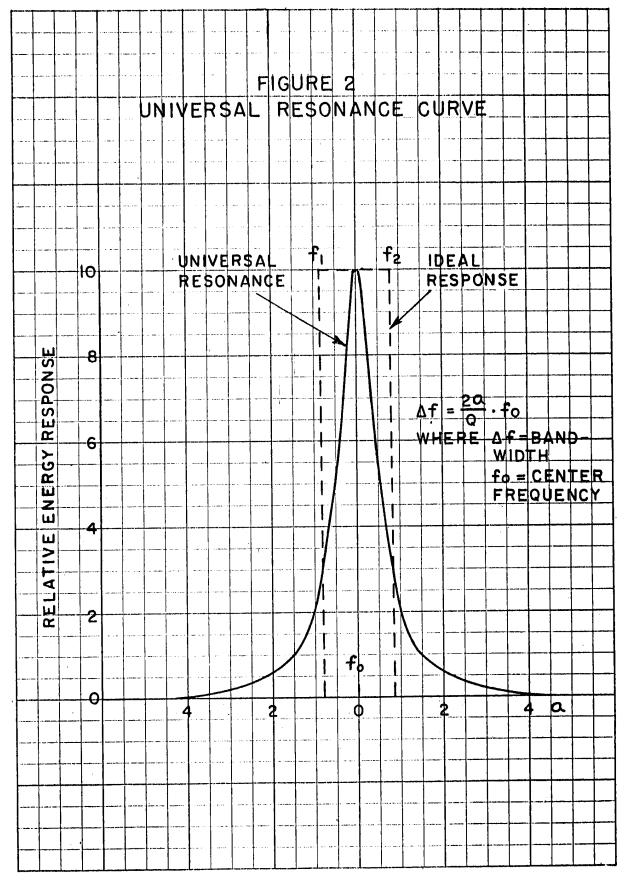
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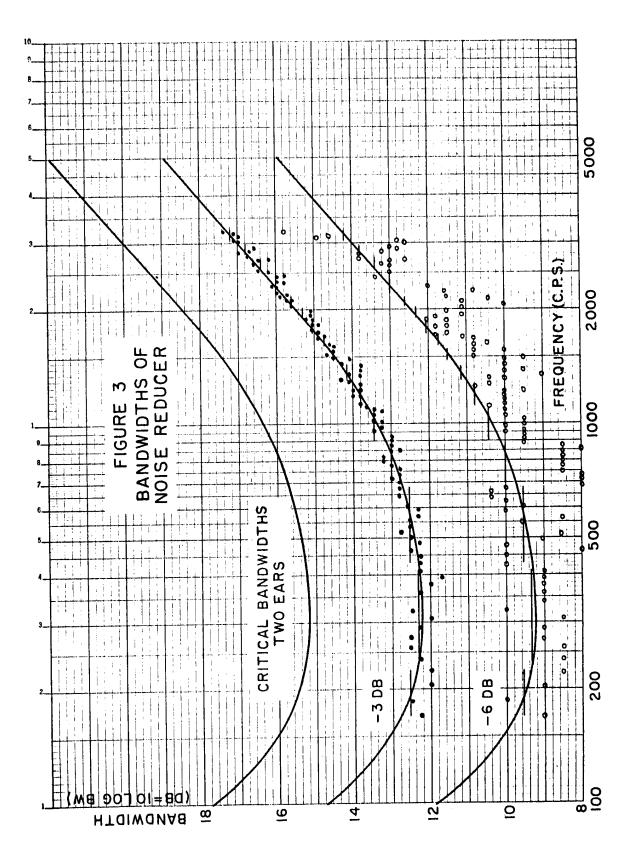
### **PERIODICALS**

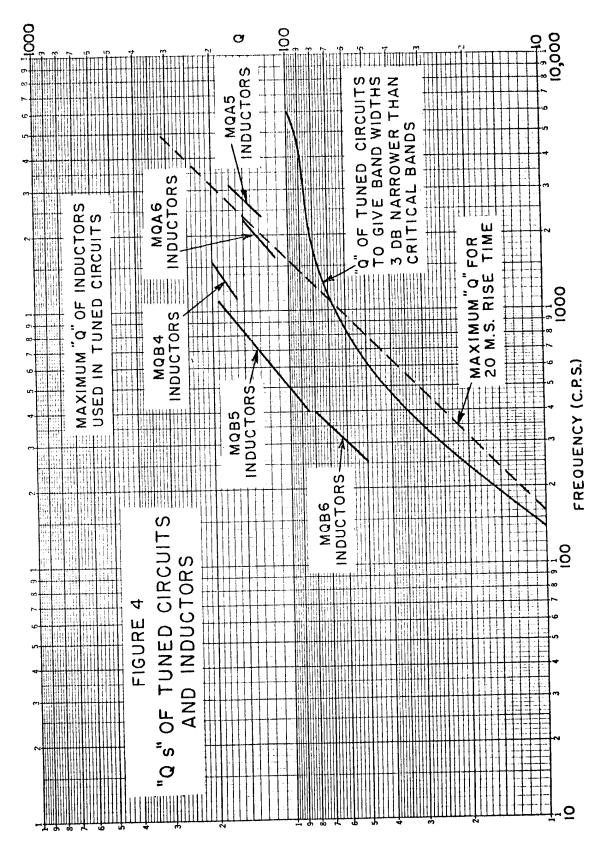
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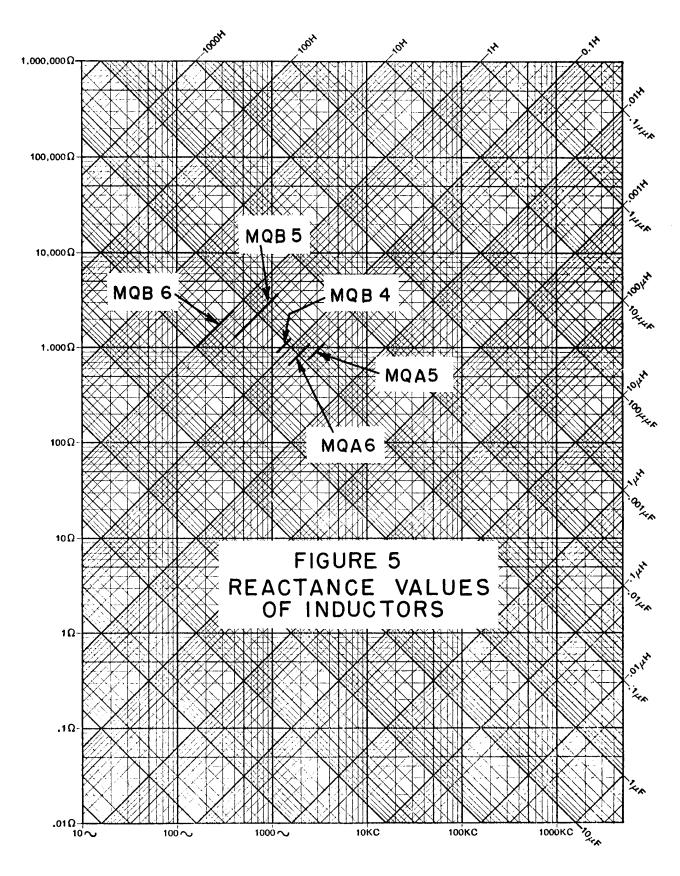
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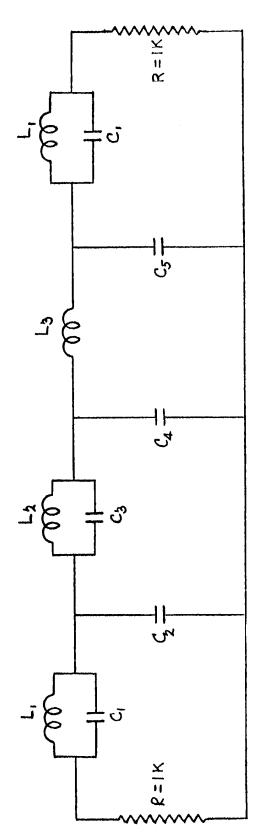












C S	.772	•515	.340	.253	961.	.158	.129	.107	060*	.075	290
2 <del>4</del> , <del>7</del>	.869	.580	.384	.284	.221	.178	.145	121.	101.	• 085	10°.
ည်း နေ	.109	.073	.048	.0356	.0278	.0223	.0182	.015	.013	.0106	• 0088
F %	.675	.451	.298	.221	171.	.138	.113	₹60•	6.00	990*	.055
F J	.515	.343	.227	.168	.130	.105	•086	.071	090*	•050	.042
L <sub>3</sub>	965	642	428	316	247	197	791	134	112	94	78
7° 4	77.2	514	343	253	197	158	129	107	8	75	63
7, 4	589	192	128	95	74	59	49	40	\$	28	24
Filter No.	41	1.8	27	~	2	'n	4	5	9	7	æ

FIG. 6-LCW PASS FILTER CONFIGURATION

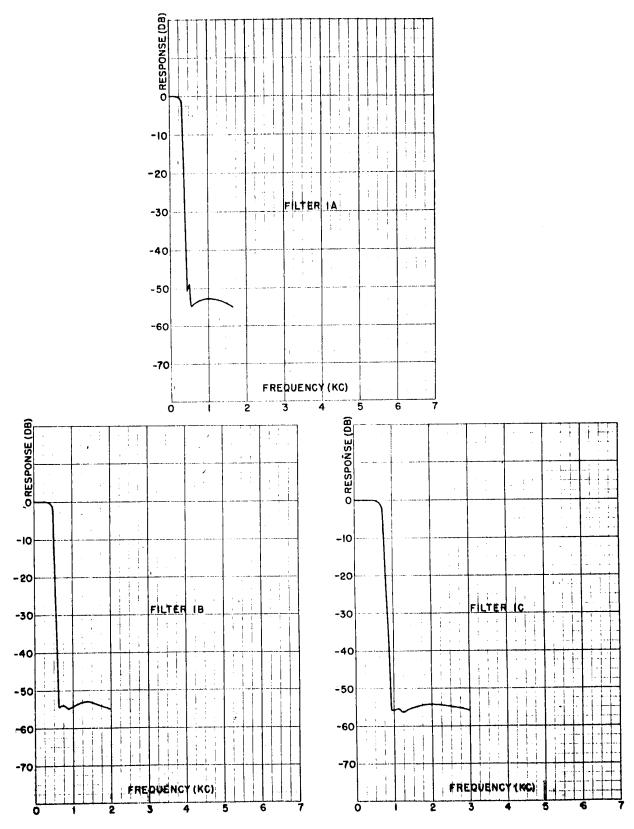
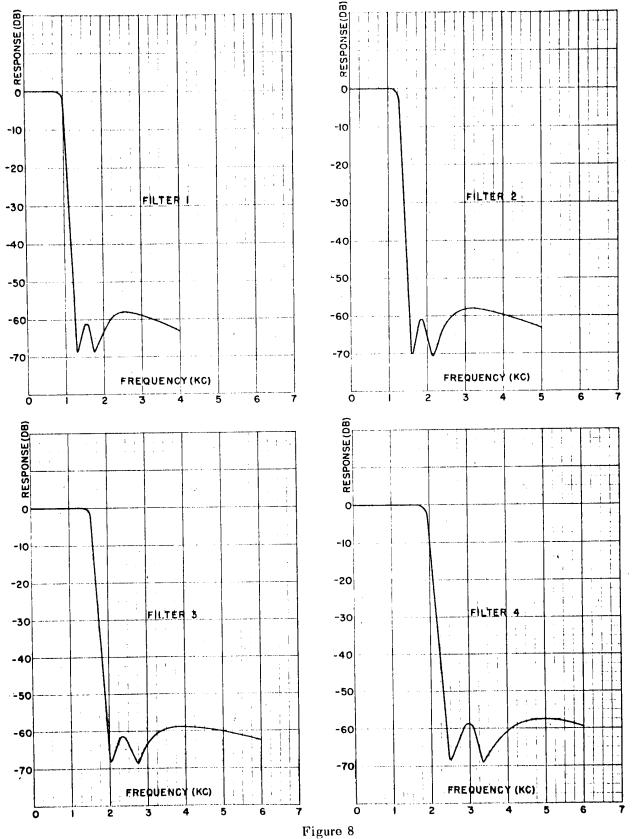


Figure 7

Response Frequency Characteristics of Low Pass Filters
38



Response Frequency Characteristics of Low Pass Filters

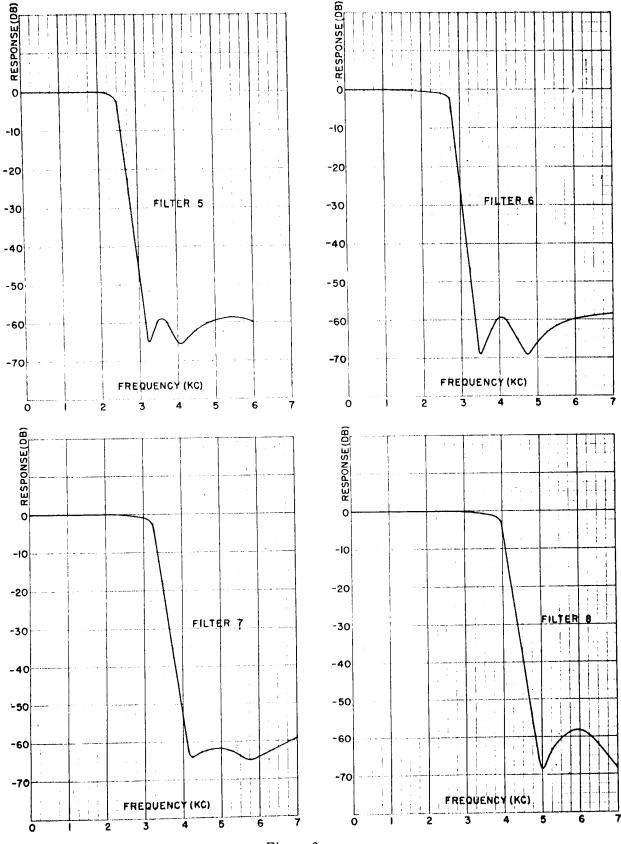
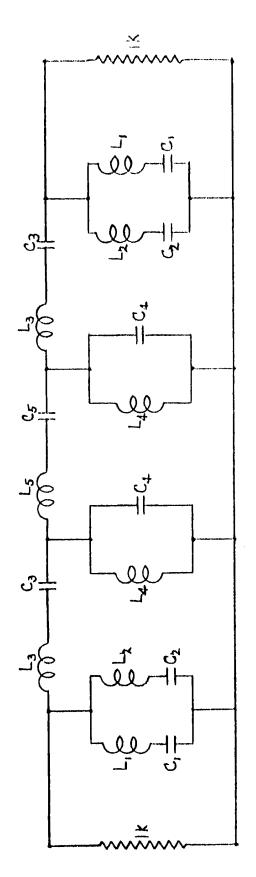
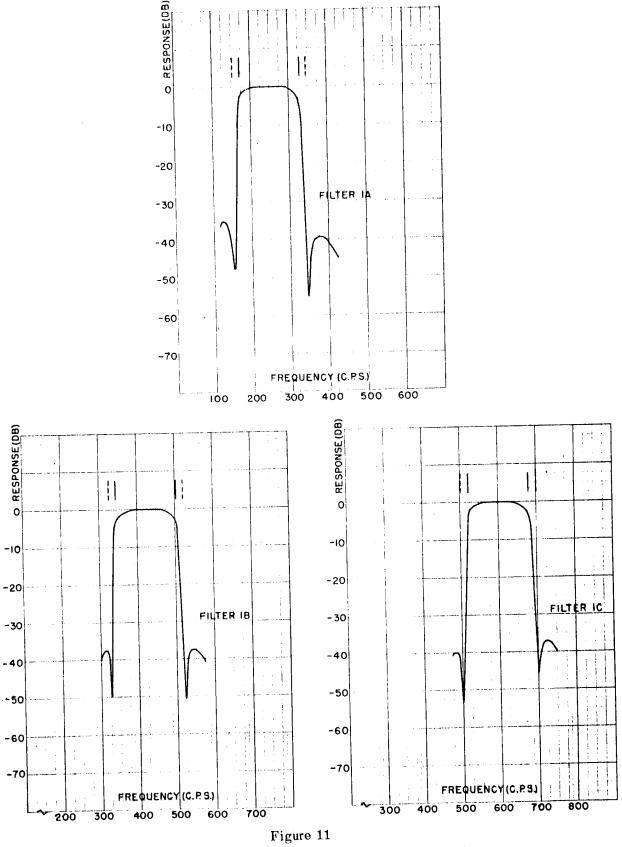


Figure 9
Response Frequency Characteristics of Low Pass Filters



								+			
<del>ار</del> ا	.248	.081	0,00	,025	.0175	.013	010.	6200.	0085	.0052	.00447
L <sub>E</sub> mb	68. 14.	1.34	1.785	1.308	1.450	1,335	1.15	1.305	.857	.753	.632
C <sub>4</sub> µ£	1.84	1,84	1.79	1,81	1.46	1.31	1.15	1.01	.857	.753	.630
L <sub>4</sub> mh	.248	.031	.0396	.0253	.0176	.013	.010	6200.	.0065	.0052	.3045
$c_{3}\mu f$	.318	.111	.054	.0345	.0244	.0184	.0142	0110	20600	.03722	. 30608
qш <sup>2</sup> -1	1.44	1,343	1,316	1,161	1.051	.920	.311	.724	.617	.542	.465
CZHE	.137	• 038	.0212	.0111	.0085	.0058	.0044	.00391	61200.	.00259	.00245
L <sub>2</sub> mh	1.652	2.59	2.50	2.38	2.426	2.4	2.152	1.70	1.464	1.280	926*
Jri¹2	.277	.0575	.0285	.0142	.0106	7,00.	.00535	.00467	.0038	.00306	.0029
L <sub>1</sub> mh	3.35	3.92	3.38	3.70	3.02	2.30	2.60	2.36	1.75	1,515	1.154
Filter	ä	13	10	1	2	3	4	ro	မ	7	8

FIGURE 10-BAND PASS FILTER CONFIGURATION



Response Frequency Characteristics of Band Pass Filters
42

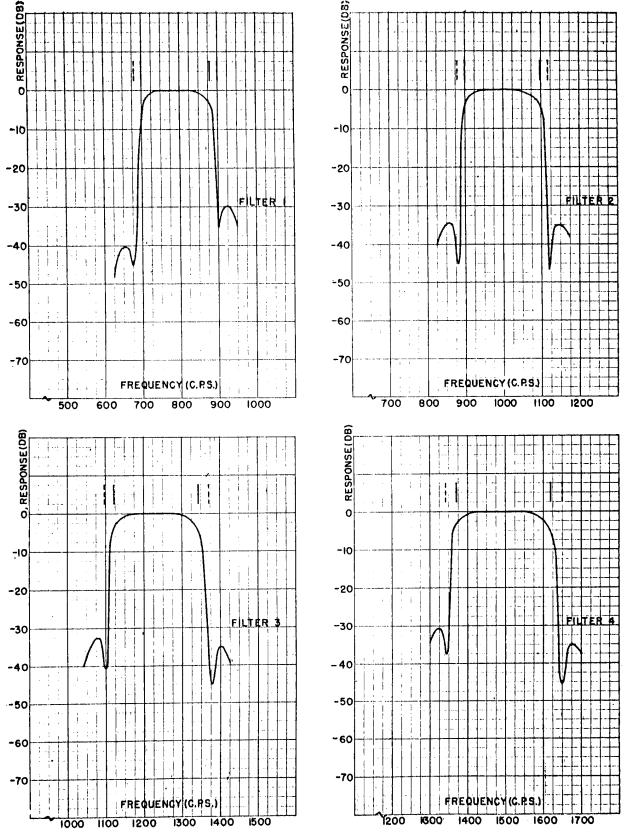
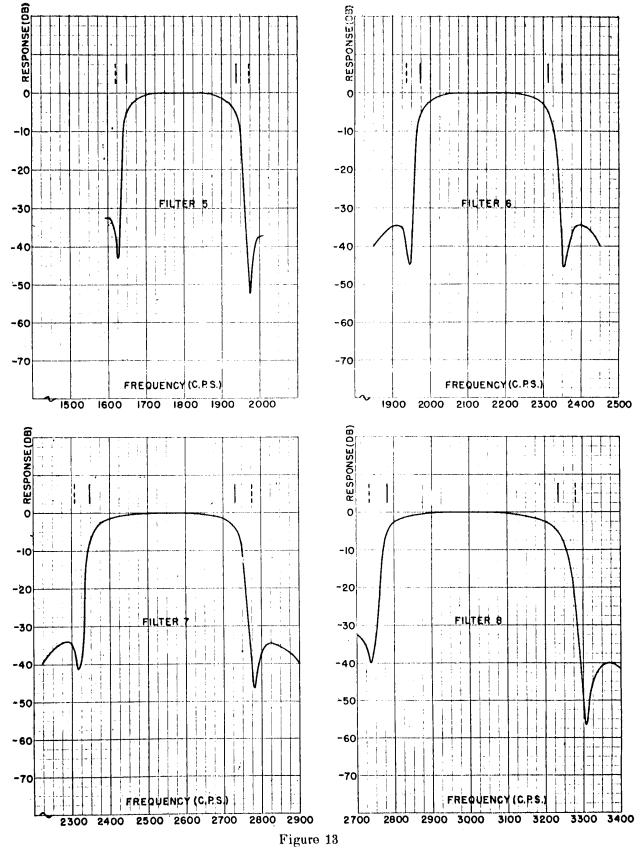
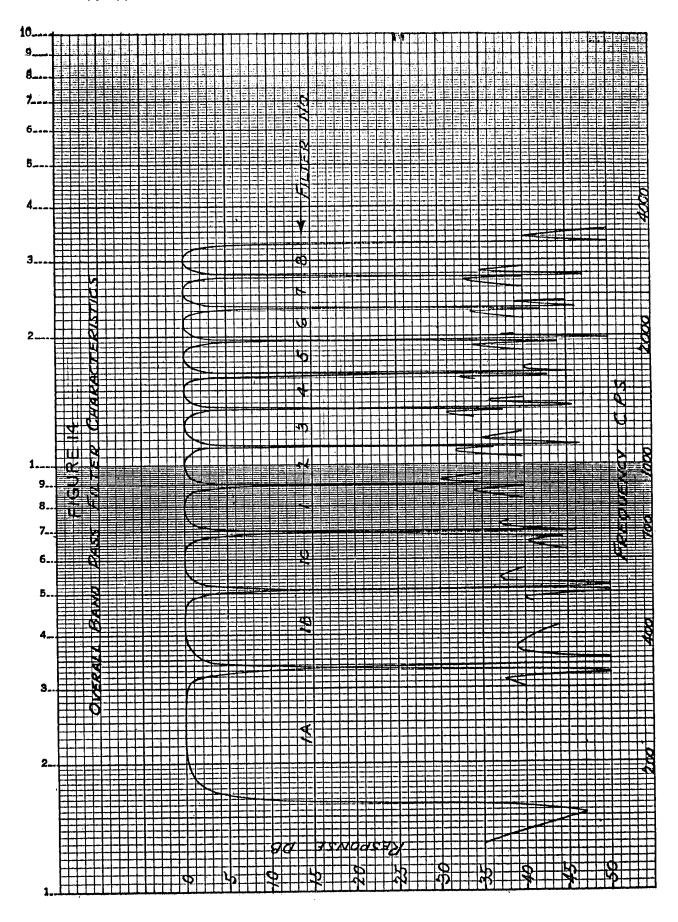


Figure 12
Response Frequency Characteristics of Band Pass Filters



Rosponse Frequency Characteristics of Band Pass Filters



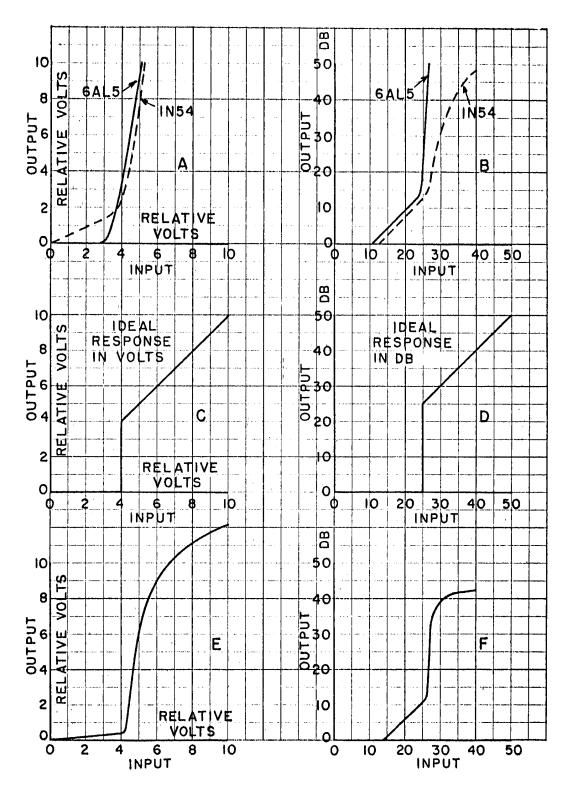
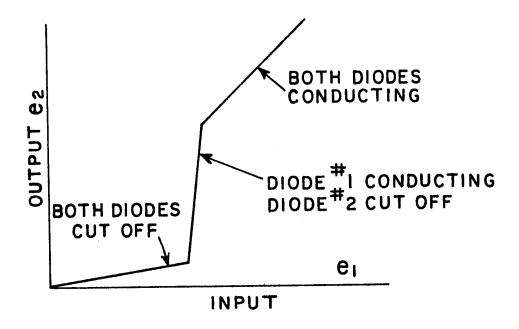
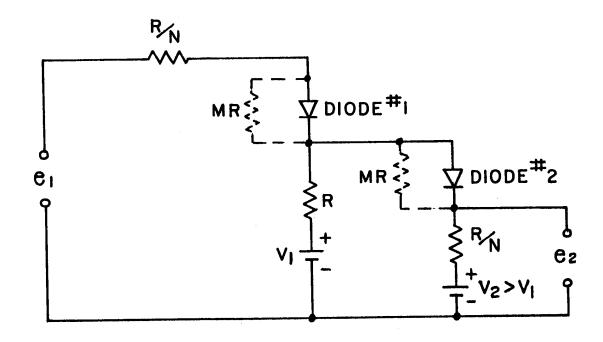


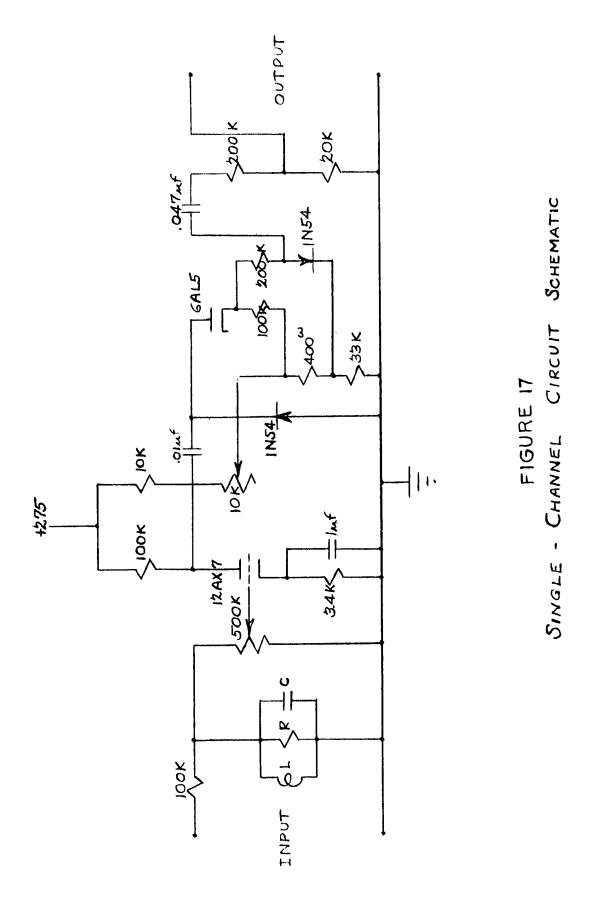
FIGURE 15
INPUT-OUTPUT CHARACTERISTICS
OF NON-LINEAR ELEMENTS



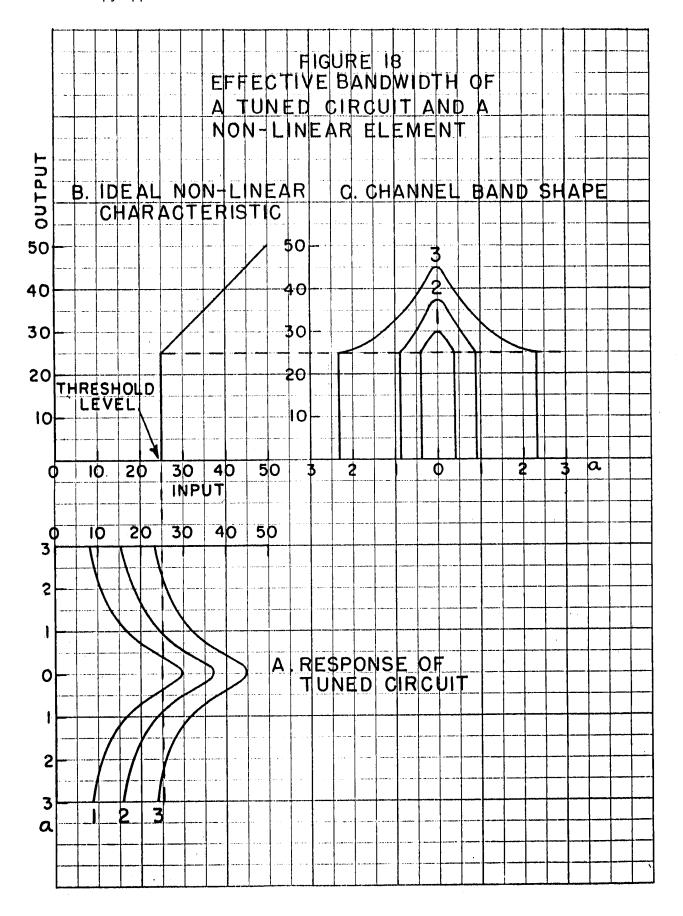


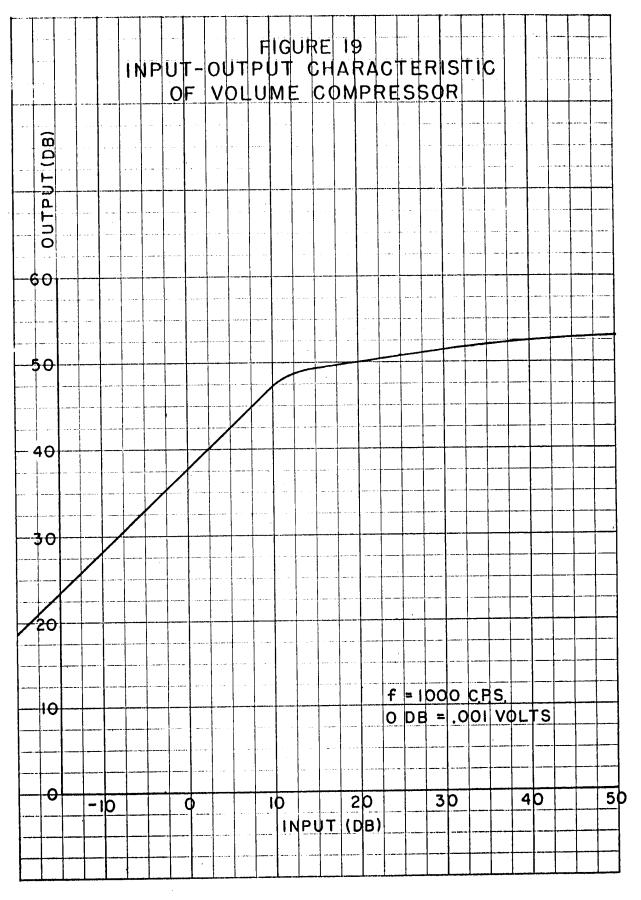
MR REPRESENTS THE DIODE LEAKAGE M AND N >> 1

FIGURE 16
NON-LINEAR ELEMENT
CIRCUIT DIAGRAM



48





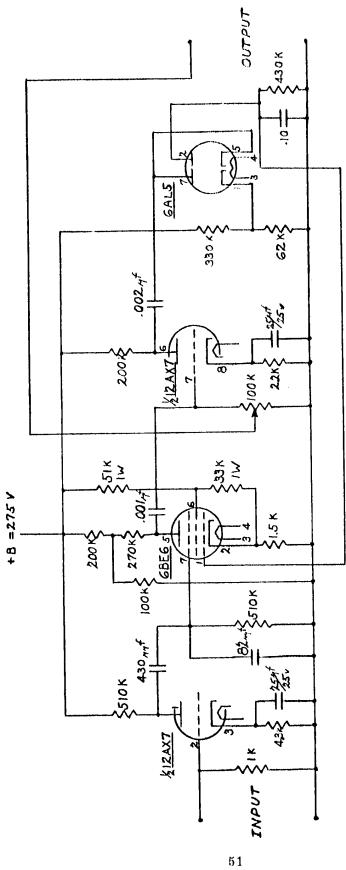
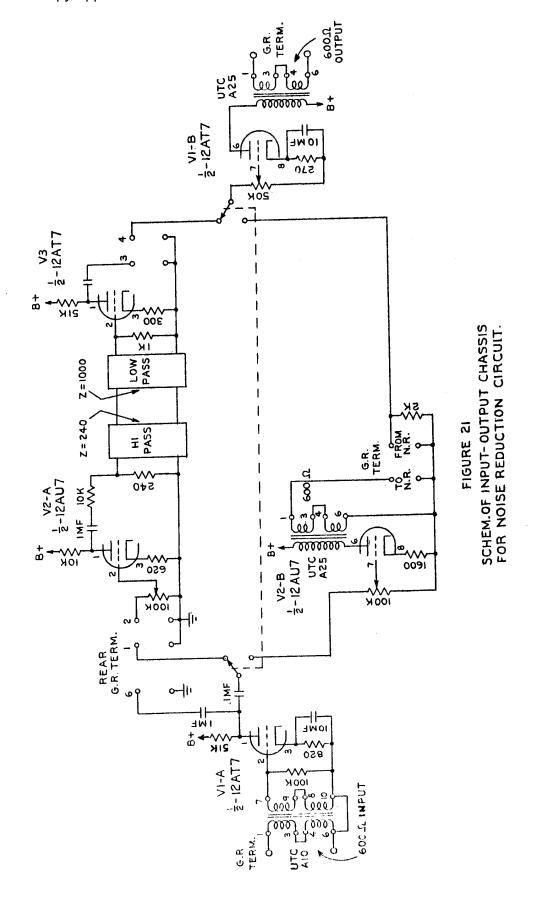
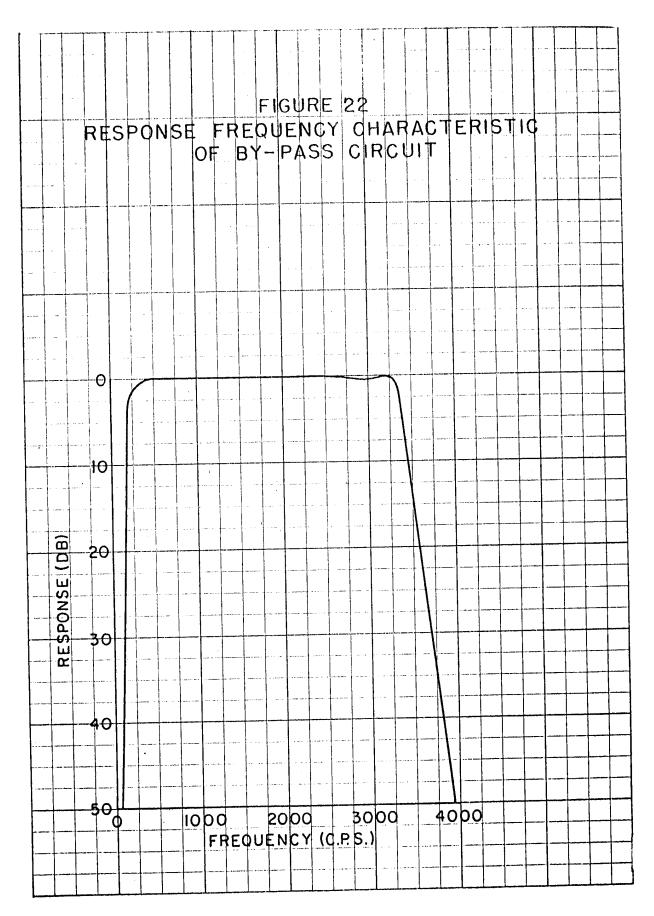
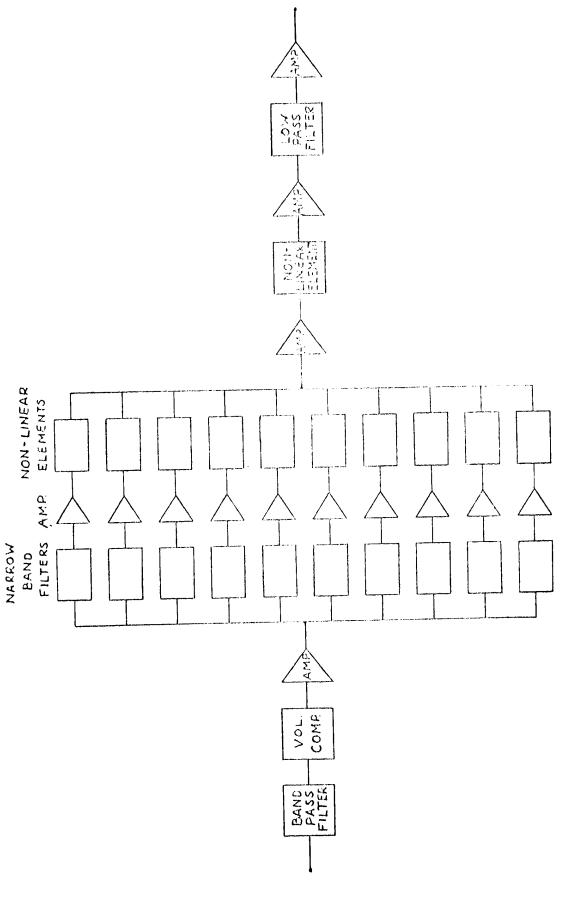


FIGURE 20 COMPRESSION







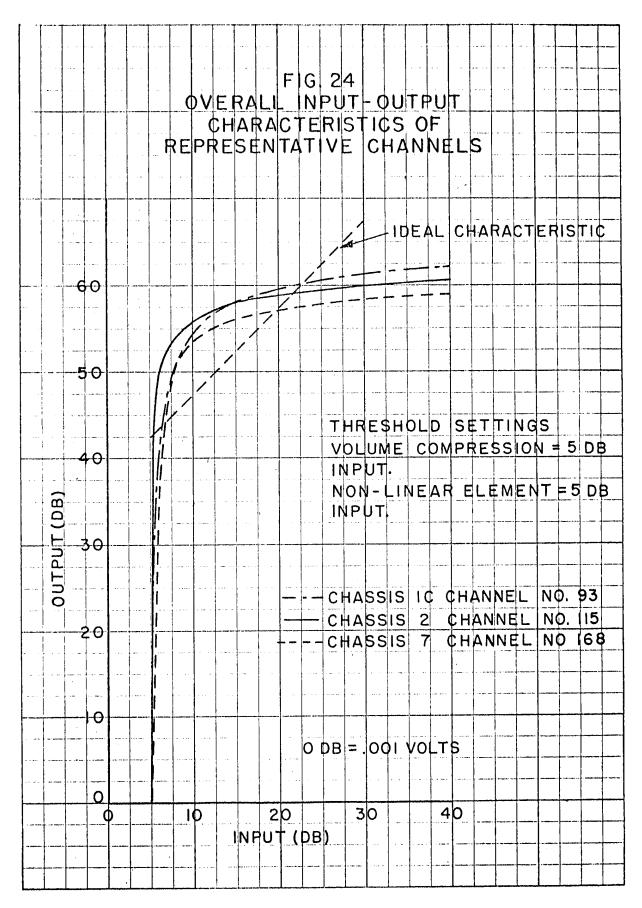
CHASSIS

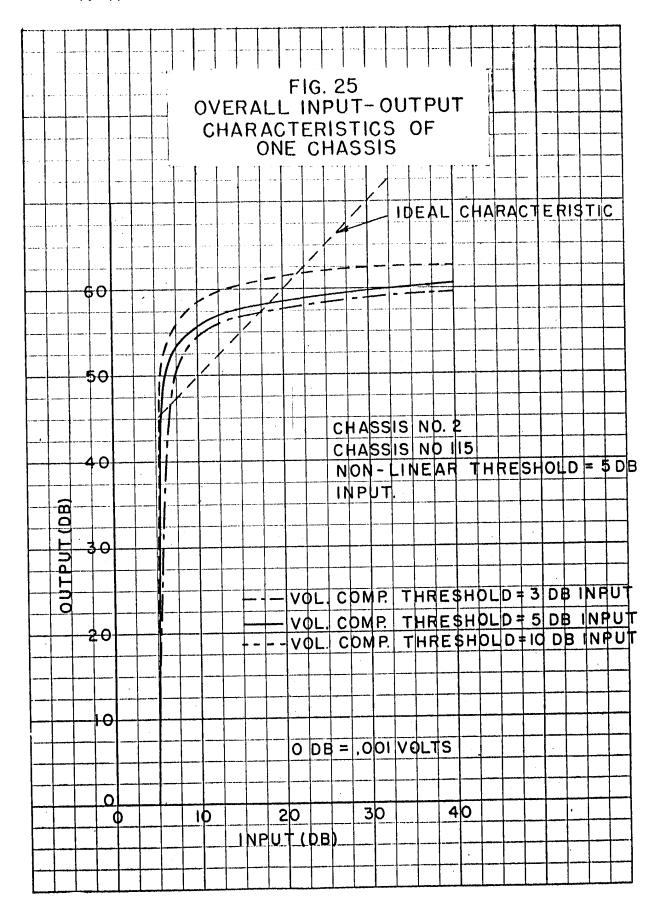
REDUCER

ONE NOISE

BLOCK DIAGRAM OF

FIGURE 23





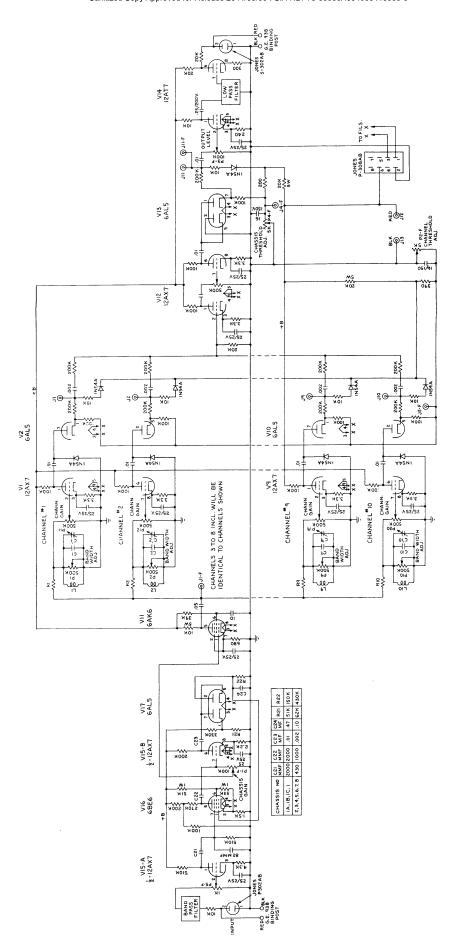
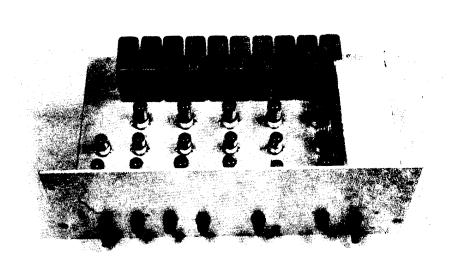


FIGURE 26. SCHEMATIC OF ONE CHASSIS FOR THRESHOLD NOISE REDUCTION CIRCUIT



10883

TOP VIEW

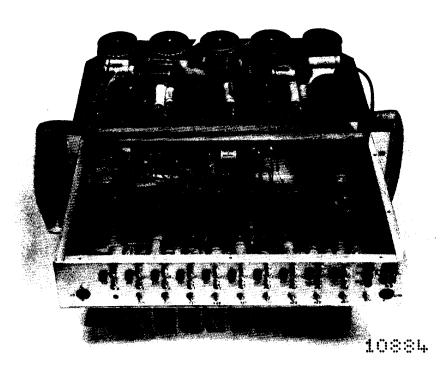
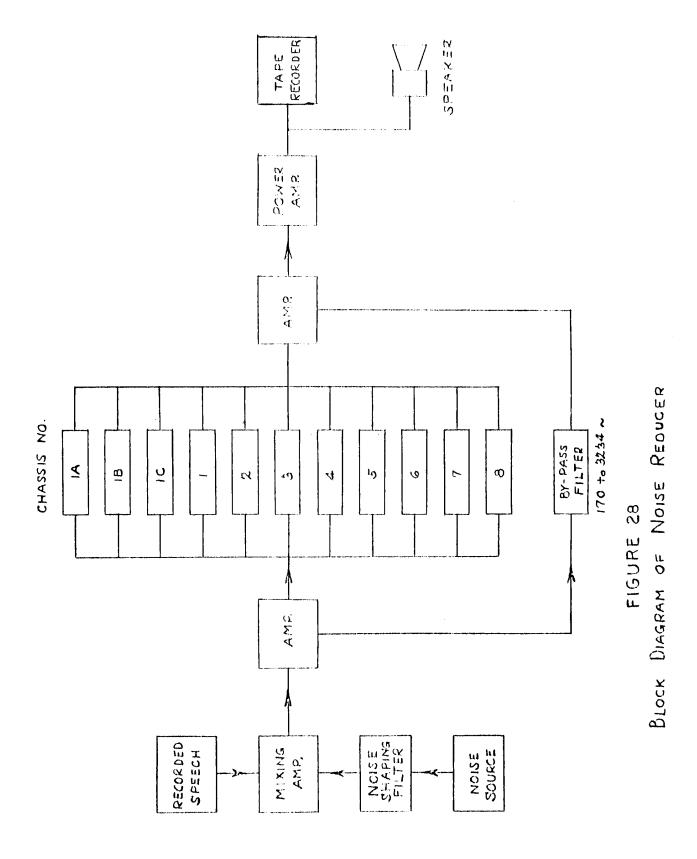


FIGURE 27
PHOTOGRAPHS OF
ONE CHASSIS

**58** 



59

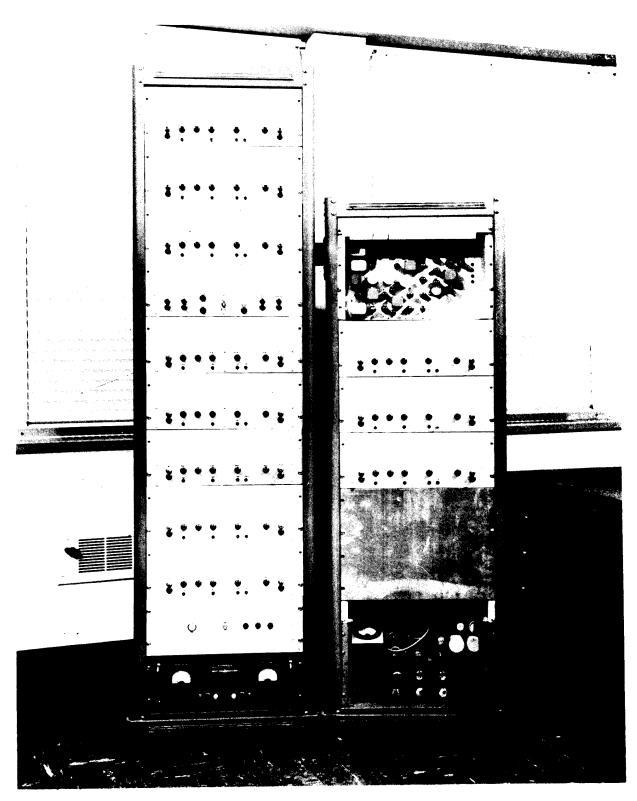


FIGURE 29
PHOTOGRAPH OF COMPLETE
CIRCUIT (FRONT VIEW)

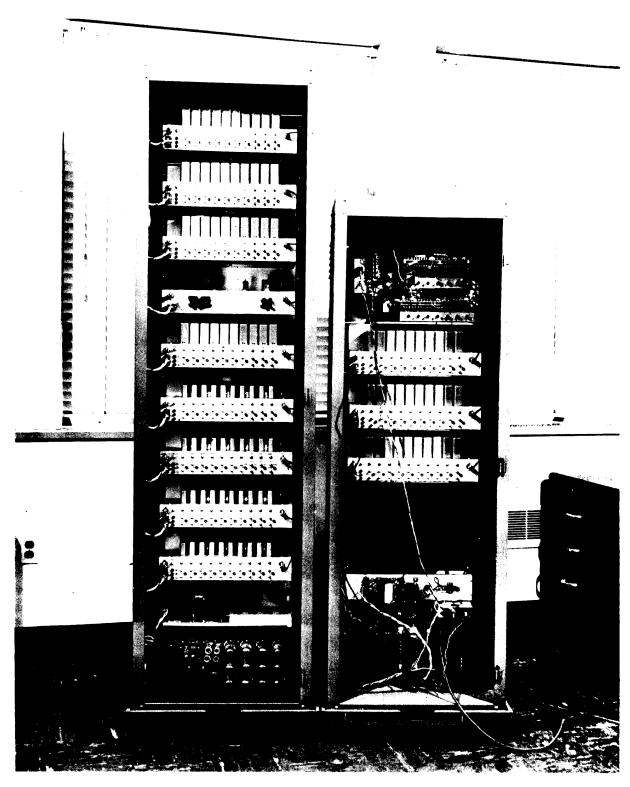


FIGURE 30
PHOTOGRAPH OF COMPLETE CIRCUIT (REAR VIEW)

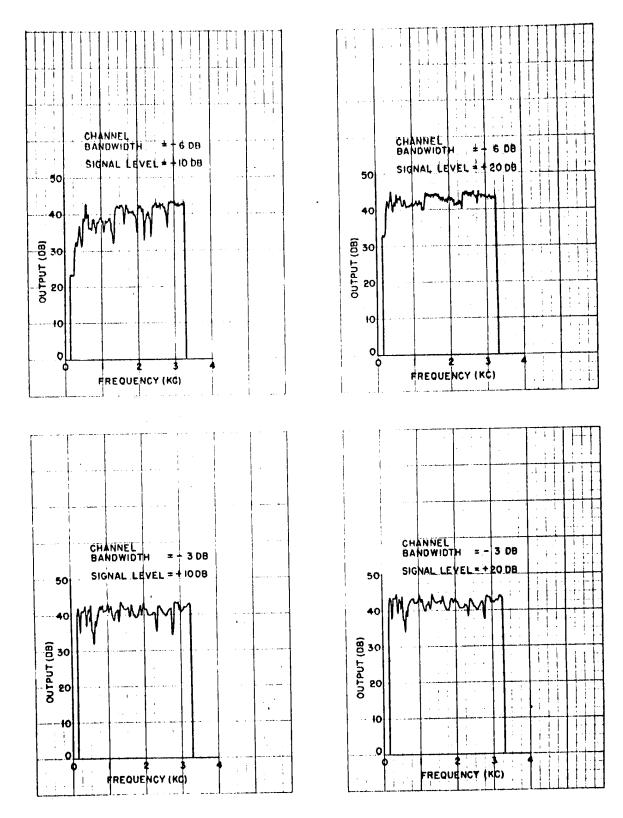
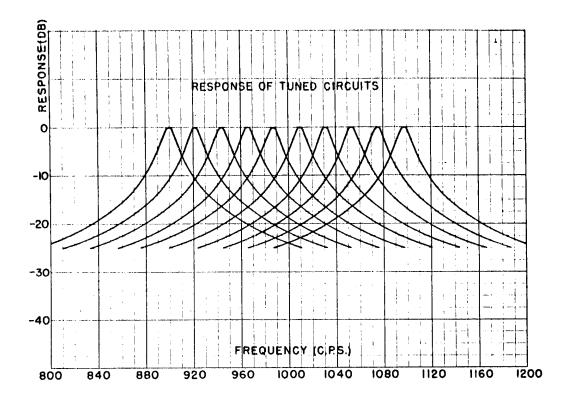
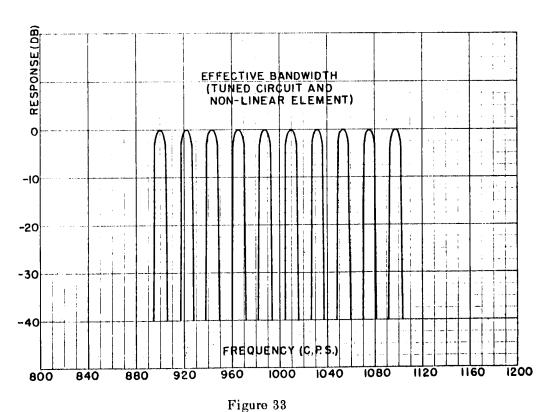
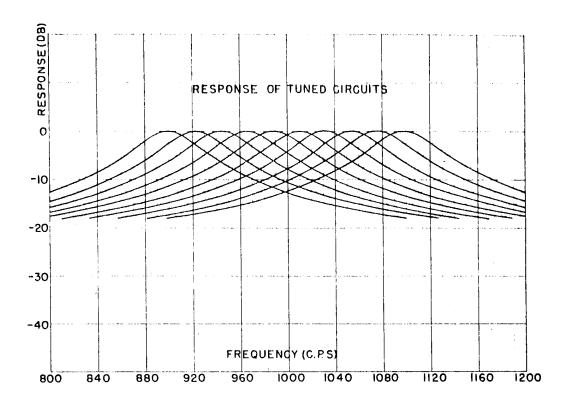


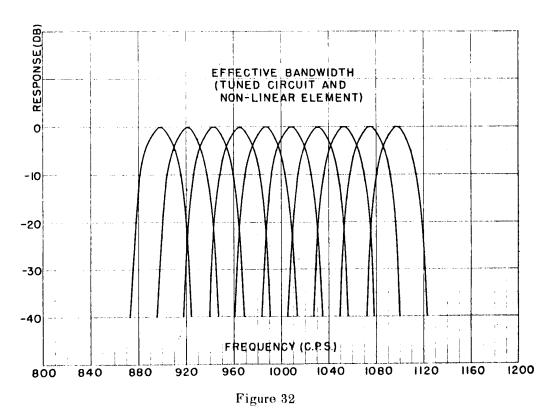
FIGURE 3I
RESPONSE FREQUENCY CHARACTERISTICS
OF COMPLETE NOISE REDUCER



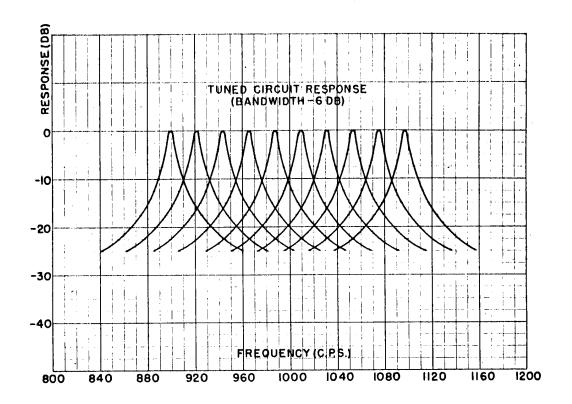


Response Frequency Characteristics of Ten Adjacent Channels
(Bandwidth -6 db)





Response Frequency Characteristics of Ten Adjacent Channels (Bandwidth -3 db)



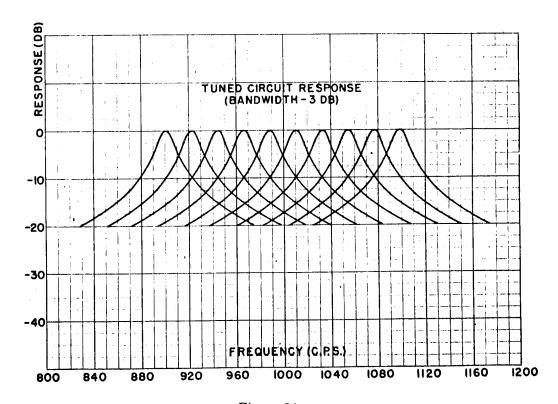
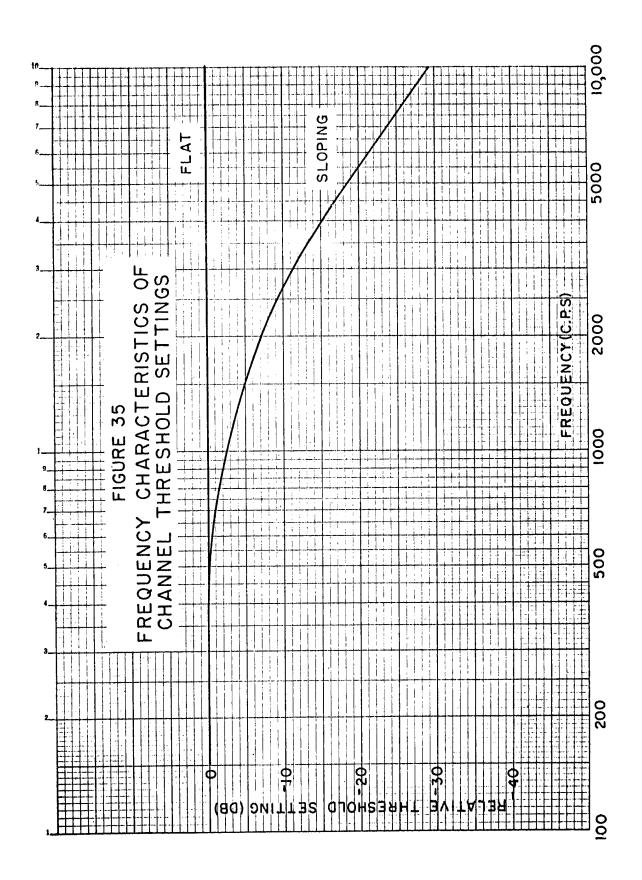
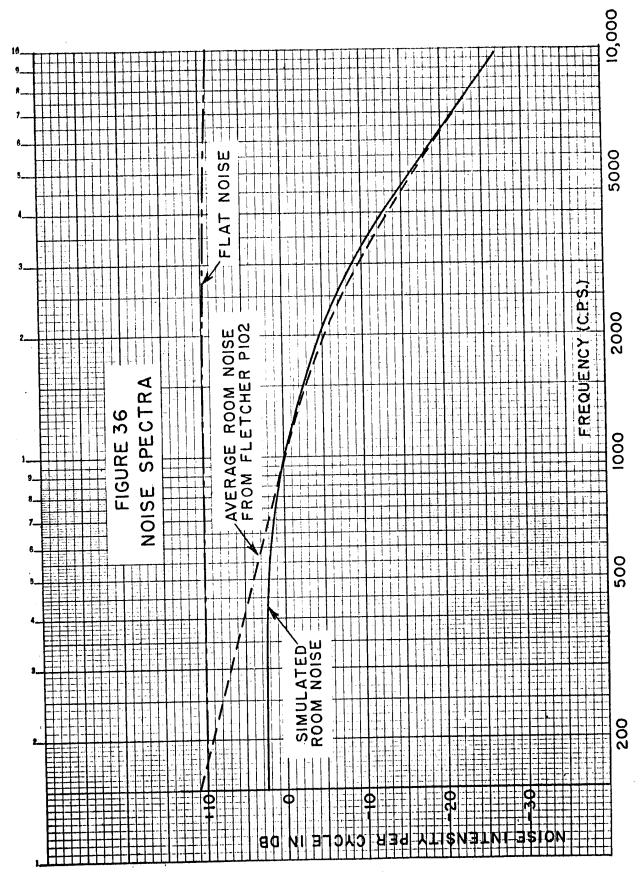
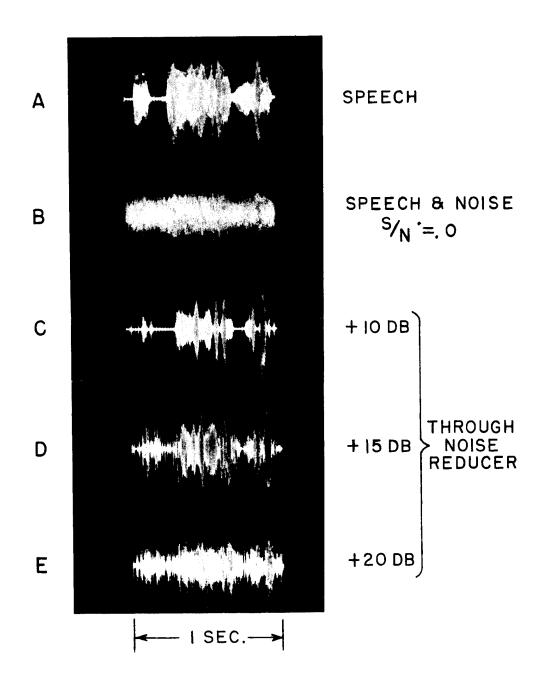


Figure 34
Response Frequency Characteristics of Ten Adjacent Channels Neglecting
Effect of Non-Linear Elements

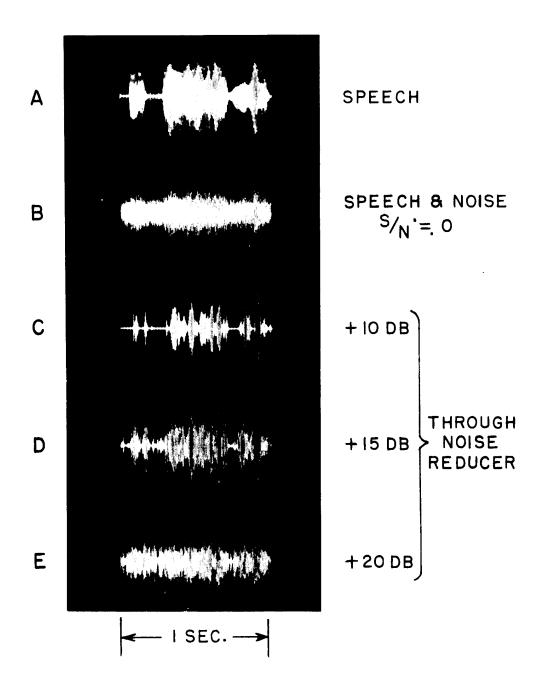






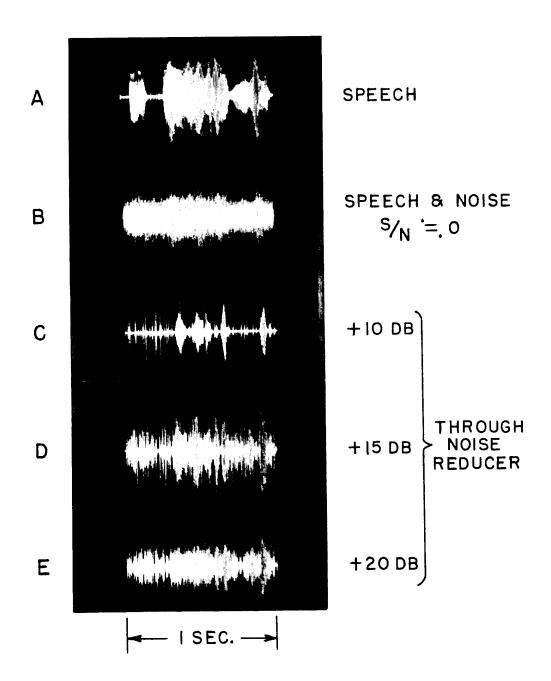
BANDWIDTH-3 DB THRESHOLD SETTING-FLAT NOISE SPECTRUM-FLAT

# FIGURE 37 OSCILLOGRAMS SHOWING EFFECTIVENESS OF NOISE REDUCER



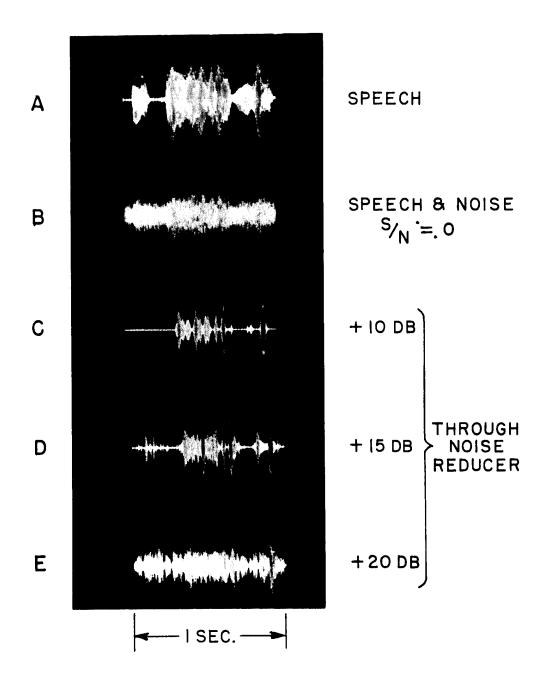
BANDWIDTH-3DB THRESHOLD SETTING-FLAT NOISE SPECTRUM-ROOM

# FIGURE 38 OSCILLOGRAMS SHOWING EFFECTIVENESS OF NOISE REDUCER



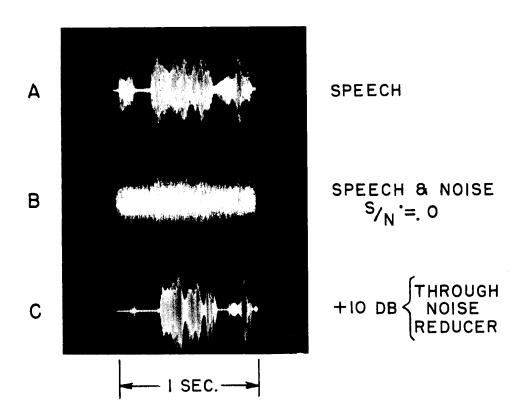
BANDWIDTH-3 DB THRESHOLD SETTING-SLOPING NOISE SPECTRUM-FLAT

# FIGURE 39 OSCILLOGRAMS SHOWING EFFECTIVENESS OF NOISE REDUCER



BANDWIDTH-3 DB THRESHOLD SETTING-SLOPING NOISE SPECTRUM-ROOM

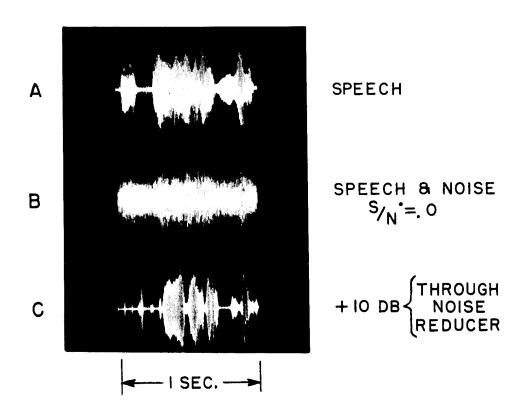
# FIGURE 40 OSCILLOGRAMS SHOWING EFFECTIVENESS OF NOISE REDUCER



BANDWIDTH - 6 DB THRESHOLD SETTING - FLAT NOISE SPECTRUM - FLAT

## FIGURE 41 OSCILLOGRAMS SHOWING EFFECTIVENESS OF NOISE REDUCER

### COMFIDENTIAL



BANDWIDTH - 6 DB THRESHOLD SETTING - FLAT NOISE SPECTRUM-ROOM

FIGURE 42
OSCILLOGRAMS SHOWING EFFECTIVENESS
OF NOISE REDUCER

